JC19 Rec'd PCT/PTO 2 1 MAY 2001

FORM PTO-1390 (Modified) (REV 11-98) U.S. DEPARTMENT OF COMMERCE PATENT AND TRADEMARK OFFICE

TRANSMITTAL LETTER TO THE UNITED STATES DESIGNATED/ELECTED OFFICE (DO/EO/US) CONCERNING A FILING UNDER 35 U.S.C. 371

ATTORNEY'S DOCKET NUMBER 112740-223	U.S. APPLICATION NO. (IF KNOWN, SEE 37 CFR 09/856399	
INTERNATIONAL APPLICATION NO. PCT/DE99/03698	INTERNATIONAL FILING DATE 19 November 1999	PRIORITY DATE CLAIMED 19 November 1998

TITLE OF INVENTION
METHOD, BASE STATION AND SUBSCRIBER STATION FOR CHANNEL CODING IN A GSM MOBILE RADIO SYSTEM

APPLICANT(S) FOR DO/EO/US
Joachim Hagenauer et al.

Applicant herewith submits to the United States Designated/Elected Office (DO/EO/US) the following items and other information:

1. ☒ This is a **FIRST** submission of items concerning a filing under 35 U.S.C. 371.
2. ☐ This is a **SECOND** or **SUBSEQUENT** submission of items concerning a filing under 35 U.S.C. 371.
3. ☒ This is an express request to begin national examination procedures (35 U.S.C. 371(f)) at any time rather than delay examination until the expiration of the applicable time limit set in 35 U.S.C. 371(b) and PCT Articles 22 and 39(1).
4. ☒ A proper Demand for International Preliminary Examination was made by the 19th month from the earliest claimed priority date.
5. ☒ A copy of the International Application as filed (35 U.S.C. 371 (c) (2))
   a. ☒ is transmitted herewith (required only if not transmitted by the International Bureau).
   b. ☐ has been transmitted by the International Bureau.
   c. ☐ is not required, as the application was filed in the United States Receiving Office (RO/US).
6. ☒ A translation of the International Application into English (35 U.S.C. 371(c)(2)).
7. ☒ A copy of the International Search Report (PCT/ISA/210).
8. ☒ Amendments to the claims of the International Application under PCT Article 19 (35 U.S.C. 371 (c)(3))
   a. ☒ are transmitted herewith (required only if not transmitted by the International Bureau).
   b. ☐ have been transmitted by the International Bureau.
   c. ☐ have not been made; however, the time limit for making such amendments has NOT expired.
   d. ☐ have not been made and will not be made.
9. ☒ A translation of the amendments to the claims under PCT Article 19 (35 U.S.C. 371(c)(3)).
10. ☐ An oath or declaration of the inventor(s) (35 U.S.C. 371 (c)(4)).
11. ☒ A copy of the International Preliminary Examination Report (PCT/IPEA/409).
12. ☐ A translation of the annexes to the International Preliminary Examination Report under PCT Article 36 (35 U.S.C. 371 (c)(5)).

Items 13 to 20 below concern document(s) or information included:

13. ☒ An Information Disclosure Statement under 37 CFR 1.97 and 1.98.
14. ☐ An assignment document for recording. A separate cover sheet in compliance with 37 CFR 3.28 and 3.31 is included.
15. ☒ A **FIRST** preliminary amendment.
16. ☐ A **SECOND** or **SUBSEQUENT** preliminary amendment.
17. ☒ A substitute specification.
18. ☐ A change of power of attorney and/or address letter.
19. ☒ Certificate of Mailing by Express Mail
20. ☒ Other items or information:

Submission of Drawings - Figure 1-6 on five sheets

PCTUS1/REV03

JC18 Rec'd PCT/PTO 21 MAY 2001

U.S. APPLICATION NO. (IF KNOWN, SEE 37 CFR 09/856399	INTERNATIONAL APPLICATION NO. PCT/DE99/03698	ATTORNEY'S DOCKET NUMBER 112740-223

21. The following fees are submitted:.

BASIC NATIONAL FEE (37 CFR 1.492 (a) (1) - (5)) :

	Fee	CALCULATIONS	PTO USE ONLY
☐ Neither international preliminary examination fee (37 CFR 1.482) nor international search fee (37 CFR 1.445(a)(2) paid to USPTO and International Search Report not prepared by the EPO or JPO	**$1,000.00**		
☒ International preliminary examination fee (37 CFR 1.482) not paid to USPTO but Internation Search Report prepared by the EPO or JPO	**$860.00**		
☐ International preliminary examination fee (37 CFR 1.482) not paid to USPTO but international search fee (37 CFR 1.445(a)(2)) paid to USPTO	**$710.00**		
☐ International preliminary examination fee paid to USPTO (37 CFR 1.482) but all claims did not satisfy provisions of PCT Article 33(1)-(4)	**$690.00**		
☐ International preliminary examination fee paid to USPTO (37 CFR 1.482) and all claims satisfied provisions of PCT Article 33(1)-(4)	**$100.00**		
ENTER APPROPRIATE BASIC FEE AMOUNT =		**$860.00**	
Surcharge of **$130.00** for furnishing the oath or declaration later than ☐ 20 ☐ 30 months from the earliest claimed priority date (37 CFR 1.492 (e)).		**$0.00**	

CLAIMS	NUMBER FILED	NUMBER EXTRA	RATE	CALCULATIONS	PTO USE ONLY
Total claims	19 - 20 =	0	x $18.00	**$0.00**	
Independent claims	3 - 3 =	0	x $80.00	**$0.00**	
Multiple Dependent Claims **(check if applicable).**			☐	**$0.00**	
	TOTAL OF ABOVE CALCULATIONS		=	**$860.00**	
Reduction of 1/2 for filing by small entity, if applicable. Verified Small Entity Statement must also be filed (Note 37 CFR 1.9, 1.27, 1.28) **(check if applicable).**			☐	**$0.00**	
		SUBTOTAL	=	**$860.00**	
Processing fee of **$130.00** for furnishing the English translation later than ☐ 20 ☐ 30 months from the earliest claimed priority date (37 CFR 1.492 (f)).			+	**$0.00**	
		TOTAL NATIONAL FEE	=	**$860.00**	
Fee for recording the enclosed assignment (37 CFR 1.21(h)). The assignment must be accompanied by an appropriate cover sheet (37 CFR 3.28, 3.31) **(check if applicable).**			☐	**$0.00**	
		TOTAL FEES ENCLOSED	=	**$860.00**	
				Amount to be: refunded	$
				charged	$

☒ A check in the amount of **$860.00** to cover the above fees is enclosed.

☐ Please charge my Deposit Account No. in the amount of to cover the above fees. A duplicate copy of this sheet is enclosed.

☒ The Commissioner is hereby authorized to charge any fees which may be required, or credit any overpayment to Deposit Account No. **02-1818** A duplicate copy of this sheet is enclosed.

NOTE: Where an appropriate time limit under 37 CFR 1.494 or 1.495 has not been met, a petition to revive (37 CFR 1.137(a) or (b)) must be filed and granted to restore the application to pending status.

SEND ALL CORRESPONDENCE TO:

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WEV

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39, 056

REGISTRATION NUMBER

May 21, 2001

DATE

BOX PCT

IN THE UNITED STATES ELECTED/DESIGNATED OFFICE
OF THE UNITED STATES PATENT AND TRADEMARK OFFICE
UNDER THE PATENT COOPERATION TREATY-CHAPTER I

PRELIMINARY AMENDMENT

APPLICANTS:	Joachim Hagenauer et al.	DOCKET NO: 112740-223	
SERIAL NO:		GROUP ART UNIT:	
		EXAMINER:	
INTERNATIONAL APPLICATION NO:		PCT/DE99/03698	
INTERNATIONAL FILING DATE:		19 November 1999	
INVENTION:	METHOD, BASE STATION AND SUBSCRIBER STATION FOR CHANNEL CODING IN A GSM MOBILE RADIO SYSTEM		

Assistant Commissioner for Patents,
Washington, D.C. 20231

Sir:

Please amend the above-identified International Application before entry into the National stage before the U.S. Patent and Trademark Office under 35 U.S.C. §371 as follows:

In the Specification:

Please replace the Specification of the present application, including the Abstract, with the following Substitute Specification:

SPECIFICATION

TITLE

METHOD, BASE STATION AND SUBSCRIBER STATION FOR CHANNEL CODING IN A GSM MOBILE RADIO SYSTEM

BACKGROUND OF THE INVENTION

Field of the Invention

The present invention relates, generally, to a method, base station and subscriber station for channel coding in a GSM mobile radio system and, more

specifically, to such a method, base station and subscriber station which uses recursive systematic codes (RSC codes) for the associated channel coding.

Description of the Prior Art

The GSM (global system for mobile communications) mobile radio system is installed in more than 100 networks and for more than 100 million subscribers worldwide. In the GSM mobile radio system, data (for example, voice or data within data services such as SMS or GPRS) are transmitted via a radio interface with the aid of electromagnetic waves. The radio interface relates to a connection between a base station and subscriber stations where the subscriber stations can be mobile stations or stationary radio stations. The electromagnetic waves are radiated in this case with carrier frequencies which are within the frequency bands of 900, 1800, and 1900 MHz in the GSM mobile radio system.

In mobile radio systems, channel coding is required for transmitting the data via the radio interface. This channel coding differs for different services, e.g. 14.4 kbps data, FR (full-rate) voice, HR (half-rate) voice. The channel coding and the complementary channel decoding at the receiving end have the aim here of achieving the lowest possible bit error rate (BER).

Hitherto, only nonsystematic nonrecursive convolutional codes (NSC - nonsystematic convolutional codes) have been used for channel coding in the GSM mobile radio system (and other comparable systems). In these codes, a coded bit is generated from a weighted sum of the current and past information bits by convolutional coding. At a coding rate of ½, e.g. 2 coded bits, which in each case come from a differently weighted sum, are thus generated from one information bit (see Figure 2). The weights in this sum, and thus the generation of the coded bits, are determined by the so-called generator polynomials. Thus, e.g., the polynomial $1 + D^3 + D^4$ determines that a coded bit is produced from the sum (XOR combination) of the current, the third last and the fourth last information bit.

The bits coded during the channel coding are transmitted via the radio interface and channel-decoded at the receiving end. A frequently used decoding algorithm is the so-called Viterbi algorithm. Since the decoding process remains the

same and is also computationally intensive, hardware chips (application-specific integrated circuits (ASICs)) are used for this purpose, especially in base stations. As a rule, these ASICs can only process a certain decoding scheme, only for nonrecursive currents in the case of GSM.

In the case of the introduction of a new voice coding message for GSM mobile radio systems, the methods hitherto proposed for the channel coding, see ETSI SMG11; Tdoc SMG11 205/98, 159/98 and 147/98, 9.28.98, are exclusively based on nonrecursive codes in order to ensure compatibility with the existing hardware which is used in millions. In spite of the involvement of many manufacturers in the development process, see Tdoc SMG11 205/98, 159/98 and 147/98, of 9.28.98, other types of code have been considered to be unusable.

The present invention, therefore, directed to a method for channel coding and corresponding devices which produce better transmission quality.

SUMMARY OF THE INVENTION

Accordingly, the present invention uses recursive systematic codes (RSC codes) for the channel coding, with voice information which is to be coded firstly being arranged on the basis of its sensitivity to transmission errors and/or on the basis of a priority which is associated with it, and being subdivided into at least first and second voice information. For first voice information, a channel coding is performed which, in a first coding step, uses error protection codes for a cyclic redundancy check and, in a second coding step, uses recursive systematic codes having a numerator polynomial and a denominator polynomial. By contrast, for second voice information, a channel coding is performed which uses recursive systematic codes having a numerator polynomial These differ from the NSC codes in that, e.g. at a rate of ½, the first "coded" bit corresponds to the current information bit (systematic) and the second coded bit is produced from the current and past information bits and past coded bits (recursive). Thus, codes which are fed back are used, making use of the fact that recursive systematic codes have distinctly better code characteristics, and thus also better characteristics with respect to the error correction, especially at high bit error rates.

The RSC codes, known from, among others, E. Offer, "Decodierung mit Qualitätsinformation bei verketteten Codiersystemen" [Decoding with quality information in concordinated coding systems], progress reports, VDI-Verlag, Series 10, Vol. 443, Düsseldorf 1996, p. 21 ff and p. 119 ff, have previously not been used since they result in changes in the decoding process and are thus not hardware-compatible. An introduction of RSC codes in the channel coding did not appear possible since the installed base stations had to be retrofitted. This is not the case, in fact, since the hardware structure can be retained both at the transmitting end and at the receiving end and, nevertheless, RSC codes can be introduced for channel decoding in the GSM mobile radio system.

It is proposed to perform post-processing on the basis of the denominator polynomial with parts of the recursive systematic code after channel decoding at the receiving end. According to an advantageous further embodiment of the present invention, the decoding process is performed as previously with decoding of a NSC code, namely the one which is identical to the nonrecursive component, the numerator polynomial, of the new RSC code. After this hardware-compatible decoding, post-processing follows in which the bits obtained this way are again coded with the denominator polynomial. This post-processing is advantageously performed via programming, that is to say in software, which can be more easily loaded into existing stations later.

The coding of the post-processing is not computationally expensive and can be performed as an additional step in every base station. This recoding provides the exact bits of the data sequence of the transmitting end.

A recursive decoding which is not possible with previously installed hardware can be replaced by decoding into two nonrecursive successive individual steps. The first step is decoding using the numerator polynomial of the recursive code and the second step is a coding using the denominator polynomial of the recursive code. This makes it possible to reproduce, if necessary, any systematic recursive codes using hardware which has already been installed. The first step corresponds to the previous decoding and the second step is the post-processing.

The polynomials of identical RSC and NSC codes will be explained briefly in connection with Figures 2 and 3. In a typical NSC code (such as, e.g. GSM/TCHFS).

The generator polynomials there are:

Polynomials of the NSC codes: $G_1 = 1 + D^3 + D^4$

$G_2 = 1 + D + D^3 + D^4$

An identical RSC code is generated by dividing; e.g., by G_1:

$G_1 = 1$

Polynomials of the RSC code:

$$G_2 = \frac{1 + D + D^3 + D^4}{1 + D^3 + D^4} \ldots.$$

These RSC codes have the advantage that lower bit error rates are possible in the case of core channels (up to a bit error rate of 10^{-4}) since the channel error rate is not exceeded due to the uncoded bits (systematic component). In contrast, the bit error rate of coded bits also can be greater than the channel error rate under very poor channel conditions.

According to an advantageous embodiment of the present invention, a priori knowledge is obtained from a previous detection at the receiving end and this a priori knowledge is used in a subsequent channel decoding. During the transmission of coded voice, a number of voice parameters, and thus bits, change only rarely. It is also possible to make predictions of the probable current value from the value these parameters had in the past. If then the received current value distinctly deviates from the predicted value, there is a high probability of a transmission error and, for example, the received value can be replaced by the predicted value.

This previous knowledge (a priori knowledge) is introduced in the channel decoder and has previously been impossible in most cases since the decoding algorithm had to be modified due to the use of non-systematic codes. As a rule, the modification was, in turn, not hardware-compatible. If RSC codes are used, this a priori knowledge can be introduced quite simply before the decoding process since

some of the received bits are uncoded. The decoding process itself does not need to be modified.

As already explained, some of the received bits are uncoded information bits. If the channel conditions are good, i.e. no transmission errors are to be expected, channel decoding can be omitted and only the information bits are used. The transmission quality can then be determined as early as before the channel decoder by advantageously evaluating information from a channel estimator. After that, a decision is made as to whether decoding is necessary or not. In subscriber stations in which the energy consumption is an essential quality criterion, an essential advantage is that the channel decoder can be switched off. This saves power. In addition, the hardware for channel decoding can be omitted altogether in applications, e.g. SMS applications for linking in telemetry services etc., in which a high transmission quality is always expected.

Due to a nonrecursive decoding followed by coding, it becomes possible to use RSC codes with the advantages described above in existing GSM mobile radio systems on existing hardware.

An exemplary embodiment of the present invention is explained in greater detail on the basis of the network structure of the known GSM mobile radio system according to Figure 1 and referring to the codes according to Figures 2 and 3.

Additional features and advantages of the present invention are described in, and will be apparent from, the Detailed Description of the Preferred Embodiments and the Drawings.

DESCRIPTION OF THE DRAWINGS

Figure 1 shows the network structure of a known GSM mobile radio system;

Figures 2 and 3 show RSC and NSC codes used in connection with the system of Figure 1;

Figure 4 shows a flow chart of the coding used pursuant to the teachings of the present invention;

Figure 5 shows polynomials used in the coding and decoding; and

Figure 6 shows a flow chart of the decoding.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENTS

The GSM mobile radio system shown in Figure 1 consists of a multiplicity of mobile switching centers MSC which are networked together and, respectively, establish access to a landline network PSTN. These mobile switching centers MSC are also connected to, in each case, at least one base station controller BSC for controlling base stations BS. Each of these base station controllers BSC, in turn, provides for a connection to at least one base station BS. An operation and maintenance center OMC implements control and maintenance functions for the mobile radio system or for parts thereof, respectively.

A base station BS can set up a connection to subscriber stations, e.g. mobile stations MS or other mobile and stationary terminals, via a radio interface. Each base station BS forms at least one radio cell. Figure 1 shows connections for transmitting user information between a base station BS and mobile stations MS.

In the coding methods shown, voice information is transmitted as user information. The bits of the voice information are divided into three classes with respect to the weighting (Class 1a, 1b and 2) in accordance with their sensitivity to errors. The most important bits (Class 1a) are additionally protected by a cyclic redundancy check (CRC) error protection coding. The bits of Classes 1a and 1b are convolutionally coded and punctured. In the AMR, the interleaving of the data after the coding is performed in accordance with the interleaving arrangements previously introduced for FR and HR.

Altogether, 14 coding methods are presented in conjunction with the AMR coder, from which a selection must be made in accordance with the transmission conditions. Of these, eight coding modes can be used in full-rate mode and six coding modes can be used in half-rate mode.

Trans-mission mode	Channel coding mode	Source encoding bit rate, voice	Net bit rate, in-band signaling	Channel coding bit rate, voice	Channel coding bit rate, in-band
	CH0-FS	12.20 kbit/s (GSM EFR)	0.10 bit/s	10.20 kbit/s	0.30 kbit/s
	CH1-FS	10.20 kbit/s	0.10 bit/s	12.20 kbit/s	0.30 kbit/s
	CH2-FS	7.95 kbit/s	0.10 bit/s	14.45 kbit/s	0.30 kbit/s
TCH/FR	CH3-FS	7.40 kbit/s (IS-641)	0.10 bit/s	15.00 kbit/s	0.30 kbit/s
	CH4-FS	6.70 kbit/s	0.10 bit/s	15.70 kbit/s	0.30 kbit/s
	CH5-FS	5.90 kbit/s	0.10 bit/s	16.50 kbit/s	0.30 kbit/s
	CH6-FS	5.15 kbit/s	0.10 bit/s	17.25 kbit/s	0.30 kbit/s
	CH7-FS	4.75 kbit/s	0.10 bit/s	17.65 kbit/s	0.30 kbit/s
	CH8-HS	7.95 kbit/s	0.10 bit/s	3.25 kbit/s	0.10 kbit/s
TCH/HR	CH9-HS	7.40 kbit/s (IS-41)	0.10 bit/s	3.80 kbit/s	0.10 kbit/s
	CH10-HS	6.70 kbit/s	0.10 bit/s	4.50 kbit/s	0.10 kbit/s
	CH11-HS	5.90 kbit/s	0.10 bit/s	5.30 kbit/s	0.10 kbit/s
	CH12-HS	5.15 kbit/s	0.10 bit/s	6.05 kbit/s	0.10 kbit/s
	CH13-HS	4.75 kbit/s	0.10 bit/s	6.45 kbit/s	0.10 kbit/s

An in-band signaling with 2 bits net (4 or, respectively, 8 bits gross after coding) per frame (20 ms) is used for signaling the coding mode or for signaling the transmission quality in alternating frames. The two bits can be used for signaling four coding modes. These coding modes, which can be switched between via the in-band signaling, must be previously selected.

The following order of steps to be performed applies to all modes:

1. Information of the in-band signaling is coded with a block code;
2. The user information is sorted in accordance with their significance (class);
3. The ordered bits of the user information are coded with a systematic block code (CRC), generating words with voice and parity bits;
4. These coded bits and the rest of the Class 1 bits are convolutionally coded;
5. The coded bits are punctured in order to obtain the desired bit rate,
6. Unprotected bits are inserted into the frame with punctured data (only for half-rate mode); and
7. The bits are reordered and the coded and in-band bits are interleaved, also inserting a so-called stealing flag.

The designations used have the following significance:

k, j Numbering of the bits in data block or burst

K_x Number of bits in a block, x specifies data type

n	Numbering of the output data blocks
N	A selected data block
B	Numbering of bursts or blocks
s(k)	Voice information before sorting, $k=1...K_s$ (interface 0 in Figure 4)
d(k)	Voice information before channel coding, $k=1...K_d-1$ (interface 1 in Figure 4)
id(k)	Bits of the in-band signaling, k=0,1
ic(k)	Coded bits of the in-band signaling, k=0...3 (HR), 7 (FR)
u(k)	Data after the first coding step, $k=0,1,...K_u-1$ (block coding, CRC coding) (interface 2 in Figure 4)
c(n,k), c(k)	Data after the second coding step, $k=0,1..K_c-1$, n=0,1..N,N+1 (convolutional coding), (interface 3 in Figure 4)
i(B,k)	Interleaved data, $k=0,1..K_J-1$, $B=B_0, B_0+1, ..$
e(B,k)	Bits of a burst, k=0,1,114,115; $B=B_0, B_0+1, ..$ (interface 4 in Figure 4)

Coding in full-rate mode (FR)
Coding of the bits of the in-band signaling:

id(0,1)	ic(0..7)
00	00000000
01	10111010
10	01011101
11	11100111

Distribution of the bits to classes:

Coding mode	Number of voice bits per block	Number of Class-1 bits per block	Number of Class-1a bits per block	Number of Class-1b bits per block
CH0-FS	244	244	81	163
CH1-FS	204	204	65	139
CH2-FS	159	159	75	84
CH3-FS	148	148	61	87
CH4-FS	134	134	55	79
CH5-FS	118	118	55	63
CH6-FS	103	103	49	54
CH7-FS	95	95	39	56

There are no class 2 bits.

The essential parameters for the coder and correspondingly for each decoder are specified as follows for the first coding step:

Coding mode	Coded voice bits (K_d)	CRC-protected bits (K_{d1a})	Number of tail bits (N_{tail})	Number of bits after the first coding step (K_s=K_d+6+N_{tail})
CH0-FS	244	81	5	255
CH1-FS	204	65	5	215
CH2-FS	159	75	6	171
CH3-FS	148	61	6	160
CH4-FS	134	55	6	146
CH5-FS	118	55	6	130
CH6-FS	103	49	6	115
CH7-FS	95	39	6	107

a) Parity bits:

A 6-bit CRC (cyclic redundancy check) is used for error detection. These 6 parity bits are generated by using the following cyclic generator polynomial:

$g(D) = D^6 + D^5 + D^3 + D^2 + D^1 + 1$ for the first K_{d1a} bits of Class 1, K_{d1a} specifying the number of bits of Class 1a according to the above table. The coding with the cyclic code is performed in systematic manner:

in GF(2), the polynomials:

$d(0)D(K_{d1a}+5) + d(1)D(K_{d1a}+4) + \ldots + d(K_{d1a}-1)D^{(6)} +$

$p(0)D^{(5)} + \ldots + p(4)D + p(5)$

where p(0), p(1) ... p(5) are the parity bits which, divided by g(D), give “0”.

b) Tailing bits and reordering

The information bits and parity bits are brought together and so-called tail bits are appended:

u(k) = d(k) for $k = 0, 1, \ldots, K_{d1a}-1$

u(k) = $p(k-K_{d1a})$ for $k = K_{d1a}, K_{d1a}+1, \ldots, K_{d1a}+5$

u(k) = d(k-6) for $k = K_{d1a}+6, K_{d1a}+7, \ldots, K_d+5$

u(k) = dependent on coding mode

Thus, the following contents are defined for each coding mode after the first coding step u(k):

CH0-FS: u(k) = d(k) for k = 0, 1, ..., 80
u(k) = p(k-81) for k = 81, 82, ..., 86
u(k) = d(k-6) for k = 87, 88, ..., 249
u(k) = to be specified for k = 250, 251, ..., 254

CH1-FS: u(k) = d(k) for k = 0, 1, ..., 64
u(k) = p(k-65) for k = 65, 66, ..., 70
u(k) = d(k-6) for k = 71, 72, ..., 209
u(k) = to be specified for k = 210, 211, ..., 214

CH2-FS: u(k) = d(k) for k = 0, 1, ..., 74
u(k) = p(k-75) for k = 75, 76, ..., 80
u(k) = d(k-6) for k = 81, 82, ..., 164
u(k) = to be specified for k = 165, 166, ..., 170

CH3-FS: u(k) = d(k) for k = 0, 1, ..., 60
u(k) = p(k-61) for k = 61, 62, ..., 66
u(k) = d(k-6) for k = 67, 68, ..., 153
u(k) = to be specified for k = 154, 155, ..., 159

CH4-FS: u(k) = d(k) for k = 0, 1, ..., 54

u(k) = p(k-55) for k = 55, 56, ..., 60

u(k) = d(k-6) for k = 61, 62, ..., 139

u(k) = to be specified for k = 140, 141, ..., 145

CH5-FS: u(k) = d(k) for k = 0, 1, ..., 54

u(k) = p(k-55) for k = 55, 56, ..., 60

u(k) = d(k-6) for k = 61, 62, ..., 123

u(k) = to be specified for k = 124, 125, ..., 129

CH6-FS: u(k) = d(k) for k = 0, 1, ..., 48

u(k) = p(k-49) for k = 49, 50, ..., 54

u(k) = d(k-6) for k = 55, 56, ..., 108

u(k) = to be specified for k = 109, 110, ..., 114

CH7-FS: u(k) = d(k) for k = 0, 1, ..., 38

u(k) = p(k-39) for k = 39, 40, ..., 44

u(k) = d(k-6) for k = 45, 46, ..., 100

u(k) = to be specified for k = 101, 102, ..., 106

Convolutional coder

The bits of the first coding step (u(k)) are coded with a recursive systematic convolutional code (see also Figure 4). The number of output bits after puncturing and repetition is 448 bits for all modes of the coding method.

Coding mode	**Generator polynomials of convolutional code**	**Coder rate**	**Number of bits received in the coder**	**Number of bits output by the coder**	**Number of punctured bits**	**Number of repeated bits**
CH0-FS	G12,G13	1/2	255	510	62	0
CH1-FS	G12,G13,G14	1/3	215	645	197	0
CH2-FS	G12,G15,G16	1/3	171	513	65	0
CH3-FS	G12,G15,G16	1/3	160	480	32	0
CH4-FS	G12,G15,G16	1/4	146	584	136	0
CH5-FS	G12,G15,G16,G17	1/4	130	520	72	0
CH6-FS	G12,G15,G16,G17	1/4	115	460	12	0
CH7-FS	G12,G15,G16,G17	1/4	107	428	19	39

Further details on coding/decoding using recursive codes were given in C. Berrou, A. Glavieux, “Near optimum error-correction coding and decoding: turbo codes” - “Reflections on the prize paper”, IEEE Inf. Theory Soc. Newsletter, vol. 48, No. 2, June 1998 and C. Berrou and A. Glavieux: “Near optimum error-correcting coding and decoding: turbo codes”, IEEE Trans. on Comm., vol. 44, pp. 1261-1271, October 1996.

The coding modes are presented in sequence:

CH0-FS:

A block of 255 bits {u(0)... u(254)} is coded at the rate 1/2, using the following polynomials:

$$G12 = 1$$

$$G13 = (1 + D^2 + D^4 + D^5) / (1 + D + D^2 + D^3 + D^5)$$

The coding with G12=1 refers to the input bit being only multiplied by 1; i.e., it is transmitted uncoded.

From each input bit, one output bit is in each case generated by the coding with G12 or, respectively, G13. These appear successively at the output of the coder.

Thus, a serial input sequence of 255 input bits results in a serial sequence of 510 coded bits {C(0)... C(509)} at the output of the coder, which is defined by:

$$C(2k) = u(k)$$

$$C(2k+1) = u(k)+u(k-2)+u(k-4)+u(k-5)+C(2k-1)+C(2k-3)+C(2k-5)+C(2k-9)$$

for k = 0, 1, ..., 254; u(k) = 0 for k<0: C(k) = 0 for k<0

The bits at the output are thus coded alternately with G12 and G13.

The code is punctured so that the following 62 coded bits: {C(4*j+1) for j = 79, 80, ..., 127)} and the bits C(363), C(379), C(395), C(411), C(427), C(443), C(459), C(475), C(491), C(495), C(499), C(503) and C(507) are not transmitted.

As a result, there is a block of 448 coded and punctured bits, P(0)...P(447) which is appended to the bits of an in-band signaling in c.

$$c(k+8) = P(k) \quad \text{for } k = 0, 1, ..., 447.$$

CH1-FS:

A block of 215 bits {u(0)... u(214)} is coded at the rate 1/3, using the following polynomials:

$$G12 = 1$$

$$G13 = (1 + D^2 + D^4 + D^5) / (1 + D + D^2 + D^3 + D^5)$$

$$G14 = (1 + D^3 + D^4 + D^5) / (1 + D + D^2 + D^3 + D^5)$$

resulting in 645 coded bits, {C(0)...C(645)} defined by:

$C(3k) = u(k)$

$C(3k+1) = u(k)+u(k-2)+u(k-4)+u(k-5)+C(3k-2)+C(3k-5)+C(3k-8)+C(3k-14)$

$C(3k+2) = u(k)+u(k-3)+u(k-4)+u(k-5)+C(3k-1)+C(3k-4)+C(3k-7)+C(3k-13)$

for k = 0, 1, ..., 214; u(k) = 0 for k<0; C(k) = 0 for k<0

The code is punctured so that the following 197 coded bits:

{C(12*j+5), C(12*j+8), C(12*j+11) for j = 0, 1, ..., 25, {C(12*j+2), C(12*j+5), C(12*j+8), C(12*j+11)

for j = 26, 27, ..., 52}

and the bits C(2), C(610), C(622), C(628), C(634), C(637), C(638), C(640), C(641), C(643) and C(644) are not transmitted.

As a result, there is a block of 448 coded and punctured bits, P(0)...P(447) which is appended to the bits of an in-band signaling in c.

$$c(k+8) = P(k) \quad \text{for } k = 0, 1, ..., 447.$$

CH2-FS:

A block of a=171 bits {u(0)... u(170)} is coded at the rate 1/3, using the following polynomials:

$G12 = 1$

$G15 = (1 + D + D^2 + D^3 + D^6) / (1 + D^2 + D^3 + D^5 + D^6)$

$G16 = (1 + D + D^4 + D^6) / (1 + D^2 + D^3 + D^5 + D^6)$

resulting in 513 coded bits, {C(0)... C(512)} defined by:

C(3k) = u(k)

C(3k+1) = u(k)+u(k-1)+u(k-2)+u(k-3)+u(k-6)+C(3k-5)+ C(3k-8)+C(3k-14)+C(3k-17)

C(3k+2) = u(k)+u(k-1)+u(k-4)+u(k-6)+C(3k-4)+C(3k-7)+ C(3k-11)+C(3k-16)

for k = 0, 1, ..., 170; u(k) = 0 for k<0; C(k) = 0 for k<0

The code is punctured so that the following 65 coded bits:

{C(21*j+20) for j = 0, 1, ..., 15

C(21*j+8) C(21*j+11) C(21*j+17) C(21*j+20) for j = 16, 17, ..., 23} and the bits C(1), C(2), C(4), C(5), C(8), C(326), C(332), C(488), C(497), C(499), C(502), C(505), C(506), C(508), C(509), C(511) and C(512) are not transmitted.

As a result, there is a block of 448 coded and punctured bits, P(0)...P(447) which is appended to the bits of an in-band signaling in c.

c(k+8) = P(k) for k = 0, 1, ..., 447.

The polynomials used in modes CH5-FS, CH6-FS, CH7-FS are:

$G17 = (1 + D^2 + D^3 + D^4 + D^5 + D^6) / (1 + D^2 + D^3 + D^5 + D^6)$

The significant values for modes (CH3-FS, CH4-FS, CH5-FS, CH6-FS, CH7-FS) are:

CH3-FS:

C(3k) = u(k)

C(3k+1) = u(k)+u(k-1)+u(k-2)+u(k-3)+u(k-6)+C(3k-5)+ C(3k-8)+C(3k-14)+C(3k-17)

C(3k+2) = u(k)+u(k-1)+u(k-4)+u(k-6)+C(3k-4)+C(3k-7)+ C(3k-11)+C(3k-16)

for k = 0, 1, ..., 159; u(k) = 0 for k<0; C(k) = 0 for k<0

Bits {C(18*j+2), C(21*j+8), C(21*j+11), C(21*j+17) for j = 20, 21, ..., 25} and C(353), C(359), C(470), C(473), C(475), C(476), C(478), C(479) are not transmitted.

CH4-FS:

C(4k) = u(k)

C(4k+1) = u(k)+u(k-1)+u(k-2)+u(k-3)+u(k-6)+C(4k-7)+ C(4k-11)+C(4k-19)+C(4k-23)

C(4k+2) = u(k)+u(k-1)+u(k-4)+u(k-6)+C(4k-6)+C(4k-10)+ C(4k-18)+C(4k-22)

C(4k+3) = u(k)+u(k-2)+u(k-3)+u(k-4)+u(k-5)+u(k-6)+ C(4k-5)+C(4k-9)+C(4k-17)+C(4k-21)

for k = 0, 1, ..., 145; u(k) = 0 for k<0; C(k) = 0 for k<0

Bits {C(32*j+7), C(32*j+15), C(32*j+23), C(32*j+27) C(32*j+31) for j = 0, 1, ..., 10 C(16*j+3) C(16*j+7) C(16*j+11) C(16*j+14) C(16*j+15) for j = 22, 23, ..., 35} and bits C(2), C(3), C(11), C(331), C(566), C(570), C(578), C(579), C(581), C(582) and C(583) are not transmitted.

CH5-FS:

C(4k) = u(k)

C(4k+1) = u(k)+u(k-1)+u(k-2)+u(k-3)+u(k-6)+C(4k-7)+ C(4k-11)+C(4k-19)+C(4k-23)

C(4k+2) = u(k)+u(k-1)+u(k-4)+u(k-6)+C(4k-6)+C(4k-10)+ C(4k-18)+C(4k-22)

C(4k+3) = u(k)+u(k-2)+u(k-3)+u(k-4)+u(k-5)+u(k-6)+ C(4k-5)+C(4k-9)+C(4k-17)+C(4k-21)

for k = 0, 1, ..., 129; u(k) = 0 for k<0; C(k) = 0 for k<0

Bits {C(32*j+11), C(32*j+23), C(32*j+31) for j = 0, 1, ..., 9

C(32*j+7), C(32*j+11), C(32*j+15), C(32*j+23), C(32*j+27), C(32*j+31) for j = 10, 11, ..., 15}

and bits C(499), C(510), C(514), C(515), C(518), C(519) are not transmitted.

CH6-FS:

C(4k) = u(k)

C(4k+1) = u(k)+u(k-1)+u(k-2)+u(k-3)+u(k-6)+C(4k-7)+ C(4k-11)+C(4k-19)+C(4k-23)

C(4k+2) = u(k)+u(k-1)+u(k-4)+u(k-6)+C(4k-6)+C(4k-10)+ C(4k-18)+C(4k-22)

C(4k+3) = u(k)+u(k-2)+u(k-3)+u(k-4)+u(k-5)+u(k-6)+ C(4k-5)+C(4k-9)+C(4k-17)+C(4k-21)

for k = 0, 1, ..., 114; u(k) = 0 for k<0; C(k) = 0 for k<0

Bits

{C(16*j+11) for j = 22, 23, ..., 28} and bits C(450), C(451), C(454), C(455), C(458) are not transmitted.

CH7-FS:

C(4k) = u(k)

C(4k+1) = u(k)+u(k-1)+u(k-2)+u(k-3)+u(k-6)+C(4k-7)+ C(4k-11)+C(4k-19)+C(4k-23)

C(4k+2) = u(k)+u(k-1)+u(k-4)+u(k-6)+C(4k-6)+C(4k-10)+ C(4k-18)+C(4k-22)

C(4k+3) = u(k)+u(k-2)+u(k-3)+u(k-4)+u(k-5)+u(k-6)+ C(4k-5)+C(4k-9)+C(4k-17)+C(4k-21)

for k = 0, 1, ..., 94; u(k) = 0 for k<0; C(k) = 0 for k<0

Bits

C(1), C(2), C(3), C(6), C(7), C(11), C(367), C(383), C(399), C(407), C(415), C(418), C(419), C(421), C(422), C(423), C(425), C(426), C(427) are removed. In this block of 409 coded and punctured bits, P(0)... P(408), 39 bits are repeated:

P(409+k) = P(10+k*8)   for k = 0, 1, ..., 38

Coding in half-rate mode (HR)

Coding of the bits of the in-band signaling:

id(0,1)	ic(0..3)
00	0000
01	1001
10	0111
11	1110

Distribution of the bits to classes:

Coding mode	Number of voice bits per block	Number of Class-1 bits per block	Number of Class-1a bits per block	Number of Class-1b bits per block	Number of Class-2 bits per block
CH8-HS	159	123	67	56	36
CH9-HS	148	120	61	59	28
CH10-HS	134	110	55	55	24
CH11-HS	118	102	55	47	16
CH12-HS	103	91	49	42	12
CH13-HS	95	83	39	44	12

The essential parameters for the coder and correspondingly for each decoder are specified as follows for the first coding step:

Coding mode	Number of Class 1 bits (K_{d1})	CRC-protected bits (K_{dts})	Number of tail bits (N_{tail})	Number of output bits after the first coding step (K_u=K_{d1}+6+N_{tail})
CH8-HS	123	67	5	134
CH9-HS	120	61	5	131
CH10-HS	110	55	5	121
CH11-HS	102	55	5	113
CH12-HS	91	49	6	103
CH13-HS	83	39	6	95

The information on the parity and tail bits and on the reordering corresponding to the full-rate mode.

After the first coding step u(k), the following contents are defined for each coding mode:

CH8-HS: u(k) = d(k)   for k = 0, 1, ..., 66

u(k) = p(k-67)   for k = 67, 68, ..., 72

u(k) = d(k-6) for k = 73, 74, ..., 128

u(k) = to be specified for k = 129, 130, ..., 133

CH9-HS: u(k) = d(k) for k = 0, 1, ..., 60

u(k) = p(k-61) for k = 61, 62, ..., 66

u(k) = d(k-6) for k = 67, 68, ..., 125

u(k) = to be specified for k = 126, 127, ..., 130

CH10-HS: u(k) = d(k) for k = 0, 1, ..., 54

u(k) = p(k-55) for k = 55, 56, ..., 60

u(k) = d(k-6) for k = 61, 62, ..., 115

u(k) = to be specified for k = 116, 117, ..., 120

CH11-HS: u(k) = d(k) for k = 0, 1, ..., 54

u(k) = p(k-55) for k = 55, 56, ..., 60

u(k) = d(k-6) for k = 61, 62, ..., 107

u(k) = to be specified for k = 108, 109, ..., 112

CH12-HS: u(k) = d(k) for k = 0, 1, ..., 48

u(k) = p(k-49) for k = 49, 50, ..., 54

u(k) = d(k-6) for k = 55, 56, ..., 96

u(k) = to be specified for k = 97, 98, ..., 102

CH13-HS: u(k) = d(k) for k = 0, 1, ..., 38

u(k) = p(k-39) for k = 39, 40, ..., 44

u(k) = d(k-6) for k = 45, 46, ..., 88

u(k) = to be specified for k = 89, 90, ..., 94

Convolutional coder

The bits of the first coding step (u(k)) are coded with a recursive systematic convolutional code (see also Figure 4). The number of output bits after puncturing and repetition is 448 bits for all modes of the coding method.

Coding mode	Generator polynomials of convolutional code	Number of bits received in the coder	Coder rate	Number of bits output by the coder	Number of punctured bits
CH8-HS	G12, G13	134	1/2	268	80
CH9-HS	G12, G13	131	1/2	262	66
CH10-HS	G12, G13	121	1/2	242	42
CH11-HS	G12, G13	113	1/2	226	18
CH12-HS	G12, G15, G16	103	1/3	309	97
CH13-HS	G12, G15, G16	95	1/3	285	73

The coding modes are presented in sequence:

CH8-HS:

One block of 134 bits {u(0)...u(133)} each is coded at the rate of 1/2, using the following polynomials:

$$G12 = 1$$

$$G13 = (1 + D^2 + D^4 + D^5) / (1 + D + D^2 + D^3 + D^5)$$

resulting in 268 coded bits, {C(0)...C(267)}, defined by:

$$C(2k) = u(k)$$

$$C(2k+1) = u(k)+u(k-2)+u(k-4)+u(k-5)+C(2k-1)+C(2k-3)+C(2k-5)+C(2k-9)$$

for k = 0, 1, ..., 133; u(k) = 0 for k<0; C(k) = 0 for k<0

The code is punctured so that the following 80 coded bits:

{C(8*j+3), C(8*j+7) for j = 0, 1, ..., 21

C(8*j+3), C(8*j+5), C(8*j+7) for j = 22, 23, ..., 32)} and the bits C(1), C(265) and C(267) are not transmitted.

As a result, there is a block of 188 coded and punctured bits, P(0)...P(187) which is appended to the bits of an in-band signaling in c.

$c(k+4) = P(k)$ for $k = 0, 1, ..., 187$.

Finally, 36 Class-2 bits are appended to c

$c(192+k) = d(123+k)$ for $k = 0, 1, ..., 35$.

CH9-HS:

The 262 coded bits $\{C(0)...C(261)\}$

$C(2k) = u(k)$

$$C(2k+1) = u(k)+u(k-2)+u(k-4)+u(k-5)+C(2k-1)+C(2k-3)+C(2k-5)+C(2k-9)$$

for $k = 0, 1, ..., 130$; $u(k) = 0$ for $k<0$; $C(k) = 0$ for $k<0$

are punctured so that 66 coded bits:

$\{C(16*j+3), C(16*j+7), C(16*j+11)$ for $j = 0, 1, ..., 7$

$C(16*j+3), C(16*j+7), C(16*j+11), C(16*j+15)$ for $j = 8, 9, ..., 15)\}$ and the bits $C(1)$,

$C(221), C(229), C(237), C(245), C(249), C(253), C(257), C(259)$ and $C(261)$ are not transmitted.

A block of 196 coded and punctured bits, $P(0)...P(195)$ is appended to the bits of the in-band signaling in c:

$c(k+4) = P(k)$ for $k = 0, 1, ..., 195$.

Finally, 28 Class-2 bits are appended to c:

$c(200+k) = d(120+k)$ for $k = 0, 1, ..., 27$.

CH10-HS:

The 242 coded bits $\{C(0)...C(241)\}$:

$C(2k) = u(k)$

$$C(2k+1) = u(k)+u(k-2)+u(k-4)+u(k-5)+C(2k-1)+C(2k-3)+C(2k-5)+C(2k-9)$$

for $k = 0, 1, ..., 106$; $u(k) = 0$ for $k<0$; $C(k) = 0$ for $k<0$

are punctured so that 42 coded bits:

$\{C(8*j+3)$ for $j = 0, 1, ..., 21$

C(8*j+3), C(8*j+7) for j = 22, 23, ..., 29)} and the bits C(1), C(233), C(237) and C(241) are not transmitted.

A block of 200 coded and punctured bits, P(0)...P(199) is appended to the bits of the in-band signaling in c:

c(k+4) = P(k) for k = 0, 1, ..., 199.

Finally, 24 Class-2 bits are appended to c:

c(204+k) = d(110+k) for k = 0, 1, ..., 23.

CH11-HS:

The 226 coded bits {C(0)...C(225)}:

C(2k) = u(k)

C(2k+1) = u(k)+u(k-2)+u(k-4)+u(k-5)+C(2k-1)+C(2k-3)+ C(2k-5)+C(2k-9)

for k = 0, 1, ..., 112; u(k) = 0 for k<0; C(k) = 0 for k<0 are punctured so that 18 coded bits:

{C(28*j+15) for j = 0, 1, ..., 7} and bits C(1), C(3), C(7), C(197), C(213), C(215), C(217), C(221), C(223) and C(225) are not transmitted.

A block of 208 coded and punctured bits, P(0)...P(207) is appended to the bits of the in-band signaling in c:

c(k+4) = P(k) for k = 0, 1, ..., 207.

Finally, 16 Class-2 bits are appended to c:

c(212+k) = d(96+k) for k = 0, 1, ..., 15.

CH12-HS:

The 309 coded bits {C(0)...C(308)}:

C(3k) = u(k)

C(3k+1) = u(k)+u(k-1)+u(k-2)+u(k-3)+u(k-6)+C(3k-5)+ C(3k-8)+C(3k-14)+C(3k-17)

C(3k+2) = u(k)+u(k-1)+u(k-4)+u(k-6)+C(3k-4)+C(3k-7)+ C(3k-11)+C(3k-16)

for k = 0, 1, ..., 102; u(k) = 0 for k<0; C(k) = 0 for k<0

are punctured so that 97 coded bits:

{C(12*j+5), C(12*j+8), C(12*j+11) for j = 0, 1, ..., 15

C(12*j+2), C(12*j+5), C(12*j+8), C(12*j+11) for j = 16, 17, ..., 24) and bits C(1), C(2), C(4), C(7), C(292), C(292), C(295), C(298), C(301), C(302), C(304), C(305), C(307) and C(308) are not transmitted.

A block of 212 coded and punctured bits, P(0)...P(211) is appended to the bits of the in-band signaling in c:

c(k+4) = P(k) for k = 0, 1, ..., 211.

Finally, 12 Class-2 bits are appended to c:

c(216+k) = d(91+k) for k = 0, 1, ..., 11.

CH13-HS:

The 285 coded bits {C(0)...C(284)}:

C(3k) = u(k)

C(3k+1) = u(k)+u(k-1)+u(k-2)+u(k-3)+u(k-6)+C(3k-5)+C(3k-8)+C(3k-14)+C(3k-17)

C(3k+2) = u(k)+u(k-1)+u(k-4)+u(k-6)+C(3k-4)+C(3k-7)+C(3k-11)+C(3k-16)

for k = 0, 1, ..., 94; u(k) = 0 for k<0; C(k) = 0 for k<0

are punctured so that 73 coded bits:

{C(12*j+5), C(12*j+11) for j = 0, 1, ..., 11

C(12*j+5), C(12*j+8), C(12*j+11) for j = 12, 13, ..., 22} and bits C(1), C(2), C(4), C(7), C(8), C(14), C(242), C(254), C(266), C(274), C(277), C(278), C(280), C(281), C(283) and C(284) are not transmitted.

A block of 212 coded and punctured bits, P(0)...P(211) is appended to the bits of the in-band signaling in c:

c(k+4) = P(k) for k = 0, 1, ..., 211.

Finally, 12 Class-2 bits are appended to c:

c(216+k) = d(91+k) for k = 0, 1, ..., 11.

The polynomials of the systematic recursive code (G13 to G17) in the AMR (see Figure 5) shown were used for two reasons:

- they have optimum characteristics for the puncturing; i.e., the adaptation of the data rate to the transmission rate of the radio channel, and
- numerator or denominator polynomial are in each case also the polynomial used in the original AMR channel coding proposal (see Tdoc SMG 147/98). The necessary changes are thus minimum compared with the original proposal.

The polynomials used hitherto for voice, data and signaling information in the GSM system can also be used for the AMR channel coder with negligible restrictions in the performance. This can be done instead of the polynomials described above or as a complete alternative channel coding arrangement. The advantage lies in that the compatibility is extended further since in some cases older pre-existing hardware components in the channel decoder only allow the previous GSM polynomials to be used.

Figure 6 shows a base station BS in which, in the reception case, signals received via an antenna A are amplified in a receiver, filtered, converted to baseband and digitized.

This is followed by channel decoding (step 1), which can be done with decoding devices installed in existing base stations BS; i.e., the circuit technology can remain unchanged. This is followed by post-processing (step 2) of the decoded data which is implemented as a program. This post-processing consists of convolutional coding at a rate of 1 with the denominator polynomial of the respective rate.

As a result, this post-processing is of little complexity and is performed, for example, by an additional program in a DSP (digital signal processor).

Referring, e.g. to the rate CH0-FS, this refers to the block with 255 bits at the output of the decoder being coded with the polynomial:

$$G(D) = (1 + D + D^2 + D^3 + D^5)$$

in order to obtain the 255 original bits. The number of data bits remains constant; i.e., a current data bit at the input of this post-processing yields exactly one original bit with the aid of past input bits.

The coding and decoding methods described can be used both in base stations BS and in mobile stations MS.

Although the present invention has been described with reference to specific embodiments, those of skill in the art will recognize that changes may be made thereto without departing from the spirit and scope of the invention as set forth in the hereafter appended claims.

ABSTRACT OF THE DISCLOSURE

A method, base station and subscriber station which use recursive systematic codes (RSC codes) for channel coding in GSM mobile radio systems. In contrast to previous conceptions, these RSC codes also can be used on the basis of the hardware installed in existing GSM mobile radio systems. The RSC codes can be introduced during the introduction of an adaptive multirate coder.

In the claims:

On page 25, cancel line 1, and substitute the following left-hand justified heading therefor:

We Claim as Our Invention:

Please cancel claims 1-19, without prejudice, and substitute the following claims therefor:

20. A method for channel coding in a GSM mobile radio system, wherein the channel coding uses recursive systematic codes and is performed at a transmitting end for transmission via a radio interface between a base station and a subscriber station, the method comprising the steps of:

arranging voice information to be coded based on at least one of a sensitivity of the voice information to transmission errors and a priority associated with the voice information;

subdividing the voice information into at least first and second voice information;

performing a channel coding for the first voice information which, in a first coding step, uses error protection codes for a cyclic redundancy check and, in a second coding step, uses recursive systematic codes comprising a numerator polynomial and a denominator polynomial; and

performing a channel coding for the second voice information which uses recursive systematic codes comprising a numerator polynomial and a denominator polynomial.

21. A method for channel coding in a GSM mobile radio system as claimed in claim 20, the method further comprising the steps of:

generating the error protection codes for the cyclic redundancy check using a generator polynomial

$$g(D)=D^6+D^5+D^3+D^2D^1+1.$$

22. A method for channel coding in a GSM mobile radio system as claimed in claim 20, the method further comprising the step of:

generating the recursive systematic codes using a generating polynomial

$$g(D)=1+D+D^3+D^4/1+D^3+D^4 \text{ or}$$

$$g(D)=1+D+D^4+D^6/1+D^2+D^3+D^5+D^6.$$

23. A method for channel coding in a GSM mobile radio system as claimed in claim 20, the method further comprising the step of:

performing a channel decoding comprising successive nonrecursive individual steps at a receiving end.

24. A method for channel coding in a GSM mobile radio system as claimed in claim 23, the method further comprising the step of:

performing post-processing based on the denominator polynomial after channel decoding with the numerator polynomial.

25. A method for channel coding in a GSM mobile radio system as claimed in claim 24, wherein the post-processing is performed by a programmer.

26. A method for channel coding in a GSM mobile radio system as claimed in claim 20, the method further comprising the step of:

obtaining a priori knowledge from previous decoding at a receiving end and using the a priori knowledge in subsequent channel decoding.

27. A method for channel coding in a GSM mobile radio system as claimed in claim 20, the method further comprising the step of:

switching off completely the channel decoding in a subscriber station and using, thereafter, transmitted systematic data bits which are not channel coded.

28. A method for channel coding in a GSM mobile radio system as claimed in claim 20, the method further comprising the steps of:

determining a transmission quality during a channel estimation; and

switching the channel decoding, depending on the transmission quality, at least one of on and off.

29. A method for channel coding in a GSM mobile radio system as claimed in claim 20, the method further comprising the step of:

using the recursive systematic codes in an adaptive mutirate coder wherein the coder is selected in accordance with transmission conditions.

30. A method for channel coding in a GSM mobile radio system as claimed in claim 20, the method further comprising the step of:

using at least one polynomial of a nonrecursive systematic code previously used in the GSM mobile radio system as one of the numerator and denominator polynomials of the recursive systematic codes.

31. A base station for a GSM mobile radio system which performs, for transmission via a radio interface to a subscriber station, a channel coding which uses recursive systematic codes, comprising:

an arrangement part for arranging voice information to be coded based on at least one of a sensitivity of the voice information to transmission errors and a priority which is associated with the voice information, and for subdividing the voice information into at least first and second voice information;

a first voice channel coding part for first voice information wherein channel coding is performed which, in a first coding step, uses error protection codes for a cyclic redundancy check and, in a second coding step, uses recursive systematic codes comprising a numerator polynomial and a denominator polynomial; and

a second channel coding for the second voice information wherein channel coding is performed which uses recursive systematic codes comprising a numerator polynomial and a denominator polynomial.

32. A base station for a GSM mobile radio system as claimed in claim 31, wherein the error protection codes for the cyclic redundancy check are generated using a generator polynomial

$$g(D)=D^6+D^5+D^3+D^2+D^1+1.$$

33. A base station for a GSM mobile radio system as claimed in claim 31, wherein the recursive systematic codes are generated using a generator polynomial

$$g(D)=1+D+D^3+D^4/1+D^3+D^4 \text{ or}$$
$$g(D)=1+D+D^4+D^6/1+D^2+D^3+D^5+D^6.$$

34. A subscriber station for a GSM mobile radio system which performs, for transmission via a radio interface to a base station, a channel coding which uses recursive systematic codes, comprising:

an arrangement part for arranging voice information to be coded based on at least one of a sensitivity of the voice information to transmission errors and a priority which is associated with the voice information, and for subdividing the voice information into at least first and second voice information;

a first channel coding part for first voice information wherein channel coded is performed which, in a first coding step, uses error protection codes for a cyclic redundancy check and, in a second coding step, uses recursive systematic codes comprising a numerator polynomial and denominator polynomial; and

a second channel coding part for the second voice information wherein channel coding is performed which uses recursive systematic codes comprising a numerator polynomial and a denominator polynomial.

35. A subscriber station for a GSM mobile radio system as claimed in claim 34, wherein the error protection codes for the cyclic redundancy check are generated using a generator polynomial

$$g(D)=D^6+D^5+D^3+D^2+D^1+1.$$

36. A subscriber station for a GSM mobile radio system as claimed in claim 34, wherein the recursive systematic codes are generated using a generator polynomial

$$g(D)=1+D+D^3+D^4/1+D^3+D^4 \text{ or}$$
$$g(D)=1+D+D^4+D^6/1+D^2+D^3+D^5+D^6.$$

37. A subscriber station for a GSM mobile radio system as claimed in claim 34, further comprising:

a channel decoder which can be switched off.

38. A subscriber station for a GSM mobile radio system as claimed in claim 37, wherein the channel decoder, in a switched-off state, forwards transmitted data which is not channel coded.

REMARKS

The present amendment makes editorial changes and corrects typographical errors in the specification, which includes the Abstract, in order to conform the specification to the requirements of United States Patent Practice. No new matter is added thereby. Attached hereto is a marked-up version of the changes made to the specification by the present amendment. The attached page is captioned "**Version With Markings To Show Changes Made".**

In addition, the present amendment cancels original claims 1-19 in favor of new claims 20-38. Claims 20-38 have been presented solely because the revisions by red-lining and underlining which would have been necessary in claims 1-19 in order to present those claims in accordance with preferred United States Patent Practice would have been too extensive, and thus would have been too burdensome. The present amendment is intended for clarification purposes only and not for substantial reasons related to patentability pursuant to 35 USC §§103, 102, 103 or 112. Indeed, the cancellation of claims 1-19 does not constitute an intent on the part of the Applicants to surrender any of the subject matter of claims 1-19.

Early consideration on the merits is respectfully requested.

Respectfully submitted,

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VERSIONS WITH MARKINGS TO SHOW CHANGES MADE

In The Specification:

The Specification of the present application, including the Abstract, has been amended as follows:

SPECIFICATION

TITLE

~~**Method, base station and subscriber station for channel coding in a GSM mobile radio system**~~

METHOD, BASE STATION AND SUBSCRIBER STATION FOR CHANNEL CODING IN A GSM MOBILE RADIO SYSTEM

BACKGROUND OF THE INVENTION

~~Description~~

Field of the Invention

The present invention relates, generally, to a method, base station and subscriber station for channel coding in a GSM mobile radio system and, more specifically, to such a method, base station and subscriber station which uses recursive systematic codes (RSC codes) for the associated channel coding.

Description of the Prior Art

The GSM (global system for mobile communications) mobile radio system is installed in more than 100 networks and for more than 100 million subscribers worldwide. In the GSM mobile radio system, data (for example, voice or data within data services such as SMS or GPRS) are transmitted via a radio interface with the aid of electromagnetic waves. The radio interface relates to a connection between a base station and subscriber stations where the subscriber stations can be mobile stations or stationary radio stations. The electromagnetic waves are radiated in this case with carrier frequencies which are within the frequency bands of 900, 1800, and 1900 MHz in the GSM mobile radio system.

In mobile radio systems, channel coding is required for transmitting the data via the radio interface. This channel coding differs for different services, e.g. 14.4 kbps data, FR (full-rate) voice, HR (half-rate) voice. The channel coding and

the complementary channel decoding at the receiving end have the aim here of achieving the lowest possible bit error rate (BER).

Hitherto, only nonsystematic nonrecursive convolutional codes (NSC - nonsystematic convolutional codes) have been used for channel coding in the GSM mobile radio system (and other comparable systems). In these codes, a coded bit is generated from a weighted sum of the current and past information bits by convolutional coding. At a coding rate of ½, e.g. 2 coded bits, which in each case come from a differently weighted sum, are thus generated from one information bit (see Figure 2). The weights in this sum, and thus the generation of the coded bits, are determined by the so-called generator polynomials. Thus, e.g., the polynomial $1 + D^3 + D^4$ determines that a coded bit is produced from the sum (XOR combination) of the current, the third last and the fourth last information bit.

The bits coded during the channel coding are transmitted via the radio interface and channel-decoded at the receiving end. A frequently used decoding algorithm is the so-called Viterbi algorithm. Since the decoding process remains the same and is also computationally intensive, hardware chips (application-specific integrated circuits (ASICs)) are used for this purpose, especially in base stations. As a rule, these ASICs can only process a certain decoding scheme, only for nonrecursive currents in the case of GSM.

In the case of the introduction of a new voice coding message for GSM mobile radio systems, the methods hitherto proposed for the channel coding, see ETSI SMG11; Tdoc SMG11 205/98, 159/98 and 147/98, 9.28.98, are exclusively based on nonrecursive codes in order to ensure compatibility with the existing hardware which is used in millions. In spite of the involvement of many manufacturers in the development process, see Tdoc SMG11 205/98, 159/98 and 147/98, of 9.28.98, other types of code have been considered to be unusable.

The present invention, therefore, ~~is based on the object of specifying~~ directed to a method for channel coding and corresponding devices which produce better transmission quality. ~~This object is achieved by the method having the~~

~~features of claim 1 and the devices having the features of claims 10 and 11, respectively.~~

~~According to the invention, it is proposed to use recursive systematic codes (RSC codes) for the channel coding. These~~

SUMMARY OF THE INVENTION

~~The invention is based on the object of specifying a method for channel coding and corresponding devices which produce better transmission quality. This object is achieved by the method having the features of claim 1 and the devices having the features of claims 12 and 15, respectively.~~

~~The~~ Accordingly, the present invention ~~proposes the use of~~ uses recursive systematic codes (RSC codes) for the channel coding, with voice information which is to be coded firstly being arranged on the basis of its sensitivity to transmission errors and/or on the basis of a priority which is associated with it, and being subdivided into at least first and second voice information. For first voice information, a channel coding is performed which, in a first coding step, uses error protection codes for a cyclic redundancy check and, in a second coding step, uses recursive systematic codes ~~comprising~~ having a numerator polynomial and a denominator polynomial. By contrast, for second voice information, a channel coding is performed which uses recursive systematic codes ~~comprising~~ having a numerator polynomial These differ from the NSC codes in that, e.g. at a rate of ½, the first "coded" bit corresponds to the current information bit (systematic) and the second coded bit is produced from the current and past information bits and past coded bits (recursive). Thus, codes which are fed back are used, making use of the fact that recursive systematic codes have distinctly better code characteristics, and thus also better characteristics with respect to the error correction, especially at high bit error rates.

The RSC codes, known from, among others, E. Offer, "Decodierung mit Qualitätsinformation bei verketteten Codiersystemen" [Decoding with quality information in concordinated coding systems], progress reports, VDI-Verlag, Series 10, Vol. 443, Düsseldorf 1996, p. 21 ff and p. 119 ff, have previously not been used

since they result in changes in the decoding process and are thus not hardware-compatible. An introduction of RSC codes in the channel coding did not appear possible since the installed base stations had to be retrofitted. This is not the case, in fact, since the hardware structure can be retained both at the transmitting end and at the receiving end and, nevertheless, RSC codes can be introduced for channel decoding in the GSM mobile radio system.

It is proposed to perform post-processing on the basis of the denominator polynomial with parts of the recursive systematic code after channel decoding at the receiving end. According to an advantageous further ~~development~~ embodiment of the present invention, the decoding process is performed as previously with decoding of a NSC code, namely the one which is identical to the nonrecursive component -, the numerator polynomial -, of the new RSC code. After this hardware-compatible decoding, post-processing follows in which the bits obtained ~~by~~ this way ~~means~~ are again coded with the denominator polynomial. This post-processing is advantageously performed via programming ~~means~~, that is to say in software, which can be more easily loaded into existing stations later.

The coding of the post-processing is not computationally expensive and can be performed as an additional step in every base station. This recoding provides the exact bits of the data sequence of the transmitting end.

A recursive decoding which is not possible with previously installed hardware can be replaced by decoding into two nonrecursive successive individual steps. The first step is decoding using the numerator polynomial of the recursive code and the second step is a coding using the denominator polynomial of the recursive code. This makes it possible to reproduce, if necessary, any systematic recursive codes using hardware which has already been installed. The first step corresponds to the previous decoding and the second step is the post-processing.

The polynomials of identical RSC and NSC codes will be explained briefly ~~by means of~~ in connection with Figures 2 and 3. In a typical NSC code (such as, e.g. GSM/TCHFS).

The generator polynomials there are:

Polynomials of the NSC codes: $G_1 = 1 + D^3 + D^4$

$$G_2 = 1 + D + D^3 + D^4$$

An identical RSC code is generated by dividing~~,~~; e.g., by G_1:

$$G_1 = 1$$

Polynomials of the RSC code:

$$G_2 = \frac{1 + D + D^3 + D^4}{1 + D^3 + D^4} \ldots.$$

These RSC codes have the advantage that lower bit error rates are possible in the case of core channels (up to a bit error rate of 10^{-4}) since the channel error rate is not exceeded due to the uncoded bits (systematic component). In contrast, the bit error rate of coded bits ~~can~~ also can be greater than the channel error rate under very poor channel conditions.

According to an advantageous ~~development~~ embodiment of the present invention, a priori knowledge is obtained from a previous detection at the receiving end and this a priori knowledge is used in a subsequent channel decoding. During the transmission of coded voice, a number of voice parameters, and thus bits, change only rarely. ~~or it~~ It is also possible to make predictions of the probable current value from the value these parameters had in the past. If then the received current value distinctly deviates from the predicted value, there is a high probability of a transmission error and, for example, the received value can be replaced by the predicted value.

This previous knowledge (a priori knowledge) is introduced in the channel decoder and has previously been impossible in most cases since the decoding algorithm had to be modified due to the use of non-systematic codes. As a rule, the modification was, in turn, not hardware-compatible. If RSC codes are used, this a priori knowledge can be introduced quite simply before the decoding process since some of the received bits are uncoded. The decoding process itself does not need to be modified.

As already explained, some of the received bits are uncoded information bits. If the channel conditions are good, i.e. no transmission errors are to be

expected, channel decoding can be omitted and only the information bits are used. The transmission quality can then be determined as early as before the channel decoder by advantageously evaluating information from a channel estimator. After that, a decision is made as to whether decoding is necessary or not. In subscriber stations in which the energy consumption is an essential quality criterion, an essential advantage is that the channel decoder can be switched off. This saves power. In addition, the hardware for channel decoding can be omitted altogether in applications -, e.g. SMS applications for linking in telemetry services etc. -, in which a high transmission quality is always expected.

Due to a nonrecursive decoding followed by coding, it becomes possible to use RSC codes with the advantages described above in existing GSM mobile radio systems on existing hardware.

An exemplary embodiment of the present invention is explained in greater detail on the basis of the network structure of the known GSM mobile radio system according to Figure 1 and referring to the codes according to Figures 2 and 3.

Additional features and advantages of the present invention are described in, and will be apparent from, the Detailed Description of the Preferred Embodiments and the Drawings.

DESCRIPTION OF THE DRAWINGS

Figure 1 shows the network structure of a known GSM mobile radio system;

Figures 2 and 3 show RSC and NSC codes used in connection with the system of Figure 1;

Figure 4 shows a flow chart of the coding used pursuant to the teachings of the present invention~~,~~;

Figure 5 shows polynomials used in the coding and decoding~~,~~; and

Figure 6 shows a flow chart of the decoding.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENTS

The GSM mobile radio system shown in Figure 1 consists of a multiplicity of mobile switching centers MSC which are networked together and, respectively,

establish access to a landline network PSTN. These mobile switching centers MSC are also connected to, in each case, at least one base station controller BSC for controlling base stations BS. Each of these base station controllers BSC, in turn, provides for a connection to at least one base station BS. An operation and maintenance center OMC implements control and maintenance functions for the mobile radio system or for parts thereof, respectively.

A base station BS can set up a connection to subscriber stations, mobile stations MS or other mobile and stationary terminals, via a radio interface. Each base station BS forms at least one radio cell. Figure 1 shows connections for transmitting user information between a base station BS and mobile stations MS.

In the coding methods shown, voice information is transmitted as user information. The bits of the voice information are divided into three classes with respect to the weighting (Class 1a, 1b and 2) in accordance with their sensitivity to errors. The most important bits (Class 1a) are additionally protected by a cyclic redundancy check (CRC) error protection coding. The bits of Classes 1a and 1b are convolutionally coded and punctured. In the AMR, the interleaving of the data after the coding is performed in accordance with the interleaving arrangements previously introduced for FR and HR.

Altogether, 14 coding methods are presented in conjunction with the AMR coder, from which a selection must be made in accordance with the transmission conditions. Of these, eight coding modes can be used in full-rate mode and six coding modes can be used in half-rate mode.

Trans-mission mode	Channel coding mode	Source encoding bit rate, voice	Net bit rate, in-band signaling	Channel coding bit rate, voice	Channel coding bit rate, in-band
	CH0-FS	12.20 kbit/s (GSM EFR)	0.10 bit/s	10.20 kbit/s	0.30 kbit/s
	CH1-FS	10.20 kbit/s	0.10 bit/s	12.20 kbit/s	0.30 kbit/s
	CH2-FS	7.95 kbit/s	0.10 bit/s	14.45 kbit/s	0.30 kbit/s
TCH/FR	CH3-FS	7.40 kbit/s (IS-641)	0.10 bit/s	15.00 kbit/s	0.30 kbit/s
	CH4-FS	6.70 kbit/s	0.10 bit/s	15.70 kbit/s	0.30 kbit/s
	CH5-FS	5.90 kbit/s	0.10 bit/s	16.50 kbit/s	0.30 kbit/s
	CH6-FS	5.15 kbit/s	0.10 bit/s	17.25 kbit/s	0.30 kbit/s
	CH7-FS	4.75 kbit/s	0.10 bit/s	17.65 kbit/s	0.30 kbit/s
	CH8-HS	7.95 kbit/s	0.10 bit/s	3.25 kbit/s	0.10 kbit/s
TCH/HR	CH9-HS	7.40 kbit/s (IS-41)	0.10 bit/s	3.80 kbit/s	0.10 kbit/s
	CH10-HS	6.70 kbit/s	0.10 bit/s	4.50 kbit/s	0.10 kbit/s
	CH11-HS	5.90 kbit/s	0.10 bit/s	5.30 kbit/s	0.10 kbit/s
	CH12-HS	5.15 kbit/s	0.10 bit/s	6.05 kbit/s	0.10 kbit/s
	CH13-HS	4.75 kbit/s	0.10 bit/s	6.45 kbit/s	0.10 kbit/s

An in-band signaling with 2 bits net (4 or, respectively, 8 bits gross after coding) per frame (20 ms) is used for signaling the coding mode or for signaling the transmission quality in alternating frames. The two bits can be used for signaling four coding modes. These coding modes, which can be switched between ~~by means of~~ via the1 in-band signaling, must be previously selected.

The following order of steps to be performed applies to all modes:

1. Information of the in-band signaling is coded with a block code~~,~~;
2. The user information is sorted in accordance with their significance (class)~~,~~;
3. The ordered bits of the user information are coded with a systematic block code (CRC), generating words with voice and parity bits~~,~~;
4. These coded bits and the rest of the Class 1 bits are convolutionally coded~~,~~;
5. The coded bits are punctured in order to obtain the desired bit rate,
6. Unprotected bits are inserted into the frame with punctured data (only for half-rate mode)~~,~~; and
7. The bits are reordered and the coded and in-band bits are interleaved, also inserting a so-called stealing flag.

The designations used ~~in the [lacuna]~~ have the following significance:

k, j Numbering of the bits in data block or burst

K_x Number of bits in a block, x specifies data type

n Numbering of the output data blocks

N A selected data block

B Numbering of bursts or blocks

s(k) Voice information before sorting, $k=1...K_s$ (interface 0 in Figure 4)

d(k) Voice information before channel coding, $k=1...K_d-1$ (interface 1 in Figure 4)

id(k) Bits of the in-band signaling, k=0,1

ic(k) Coded bits of the in-band signaling, k=0...3 (HR), 7 (FR)

u(k) Data after the first coding step, $k=0,1,...K_u-1$ (block coding, CRC coding) (interface 2 in Figure 4)

c(n,k), c(k) Data after the second coding step, $k=0,1..K_c-1$, n=0,1..N,N+1 (convolutional coding), (interface 3 in Figure 4)

i(B,k) Interleaved data, $k=0,1..K_j-1$, $B=B_0$, B_0+1, ..

e(B,k) Bits of a burst, k=0,1,114,115; $B=B_0$, B_0+1, .. (interface 4 in Figure 4)

Coding in full-rate mode (FR)

Coding of the bits of the in-band signaling:

id(0,1)	**ic(0..7)**
00	00000000
01	10111010
10	01011101
11	11100111

Distribution of the bits to classes:

Coding mode	Number of voice bits per block	Number of Class-1 bits per block	Number of Class-1a bits per block	Number of Class-1b bits per block
CH0-FS	244	244	81	163
CH1-FS	204	204	65	139
CH2-FS	159	159	75	84
CH3-FS	148	148	61	87
CH4-FS	134	134	55	79
CH5-FS	118	118	55	63
CH6-FS	103	103	49	54
CH7-FS	95	95	39	56

There are no class 2 bits.

The essential parameters for the coder and correspondingly for each decoder are specified as follows for the first coding step:

Coding mode	Coded voice bits (K_d)	CRC-protected bits (K_{d1a})	Number of tail bits (N_{tail})	Number of bits after the first coding step (K_s=K_d+6+N_{tail})
CH0-FS	244	81	5	255
CH1-FS	204	65	5	215
CH2-FS	159	75	6	171
CH3-FS	148	61	6	160
CH4-FS	134	55	6	146
CH5-FS	118	55	6	130
CH6-FS	103	49	6	115
CH7-FS	95	39	6	107

a) Parity bits:

A 6-bit CRC (cyclic redundancy check) is used for error detection. These 6 parity bits are generated by using the following cyclic generator polynomial: $g(D) = D^6 + D^5 + D^3 + D^2 + D^1 + 1$ for the first K_{d1a} bits of Class 1, K_{d1a} specifying the number of bits of Class 1a according to the above table. The coding with the cyclic code is performed in systematic manner:

in GF(2), the polynomials:

$d(0)D(K_{d1a}+5) + d(1)D(K_{d1a}+4) + \ldots + d(K_{d1a}-1)D^{(6)} +$

$p(0)D^{(5)} + \ldots + p(4)D + p(5)$

where p(0), p(1) ... p(5) are the parity bits which, divided by g(D), give “0”.

b) Tailing bits and reordering

The information bits and parity bits are brought together and so-called tail bits are appended:

u(k) = d(k) for $k = 0, 1, \ldots, K_{d1a}-1$

u(k) = $p(k-K_{d1a})$ for $k = K_{d1a}, K_{d1a}+1, \ldots, K_{d1a}+5$

u(k) = d(k-6) for $k = K_{d1a}+6, K_{d1a}+7, \ldots, K_d+5$

u(k) = dependent on coding mode

Thus, the following contents are defined for each coding mode after the first coding step u(k):

CH0-FS: u(k) = d(k) for k = 0, 1, ..., 80
u(k) = p(k-81) for k = 81, 82, ..., 86
u(k) = d(k-6) for k = 87, 88, ..., 249
u(k) = to be specified for k = 250, 251, ..., 254

CH1-FS: u(k) = d(k) for k = 0, 1, ..., 64
u(k) = p(k-65) for k = 65, 66, ..., 70
u(k) = d(k-6) for k = 71, 72, ..., 209
u(k) = to be specified for k = 210, 211, ..., 214

CH2-FS: u(k) = d(k) for k = 0, 1, ..., 74
u(k) = p(k-75) for k = 75, 76, ..., 80
u(k) = d(k-6) for k = 81, 82, ..., 164
u(k) = to be specified for k = 165, 166, ..., 170

CH3-FS: u(k) = d(k) for k = 0, 1, ..., 60
u(k) = p(k-61) for k = 61, 62, ..., 66
u(k) = d(k-6) for k = 67, 68, ..., 153
u(k) = to be specified for k = 154, 155, ..., 159

CH4-FS: u(k) = d(k) for k = 0, 1, ..., 54

u(k) = p(k-55) for k = 55, 56, ..., 60

u(k) = d(k-6) for k = 61, 62, ..., 139

u(k) = to be specified for k = 140, 141, ..., 145

CH5-FS: u(k) = d(k) for k = 0, 1, ..., 54

u(k) = p(k-55) for k = 55, 56, ..., 60

u(k) = d(k-6) for k = 61, 62, ..., 123

u(k) = to be specified for k = 124, 125, ..., 129

CH6-FS: u(k) = d(k) for k = 0, 1, ..., 48

u(k) = p(k-49) for k = 49, 50, ..., 54

u(k) = d(k-6) for k = 55, 56, ..., 108

u(k) = to be specified for k = 109, 110, ..., 114

CH7-FS: u(k) = d(k) for k = 0, 1, ..., 38

u(k) = p(k-39) for k = 39, 40, ..., 44

u(k) = d(k-6) for k = 45, 46, ..., 100

u(k) = to be specified for k = 101, 102, ..., 106

Convolutional coder

The bits of the first coding step (u(k)) are coded with a recursive systematic convolutional code (see also Figure 4). The number of output bits after puncturing and repetition is 448 bits for all modes of the coding method.

Coding mode	**Generator polynomials of convolutional code**	**Coder rate**	**Number of bits received in the coder**	**Number of bits output by the coder**	**Number of punctured bits**	**Number of repeated bits**
CH0-FS	G12,G13	1/2	255	510	62	0
CH1-FS	G12,G13,G14	1/3	215	645	197	0
CH2-FS	G12,G15,G16	1/3	171	513	65	0
CH3-FS	G12,G15,G16	1/3	160	480	32	0
CH4-FS	G12,G15,G16	1/4	146	584	136	0
CH5-FS	G12,G15,G16,G17	1/4	130	520	72	0
CH6-FS	G12,G15,G16,G17	1/4	115	460	12	0
CH7-FS	G12,G15,G16,G17	1/4	107	428	19	39

Further details on coding/decoding using recursive codes were given in C. Berrou, A. Glavieux, "Near optimum error-correction coding and decoding: turbo codes" - "Reflections on the prize paper", IEEE Inf. Theory Soc. Newsletter, vol. 48, No. 2, June 1998 and C. Berrou and A. Glavieux: "Near optimum error-correcting coding and decoding: turbo codes", IEEE Trans. on Comm., vol. 44, pp. 1261-1271, October 1996.

The coding modes are presented in sequence:

CH0-FS:

A block of 255 bits {u(0)... u(254)} is coded at the rate 1/2, using the following polynomials:

$G12 = 1$

$G13 = (1 + D^2 + D^4 + D^5) / (1 + D + D^2 + D^3 + D^5)$

The coding with G12=1 ~~means that~~ refers to the input bit ~~is~~ being only multiplied by 1~~,~~; i.e., it is transmitted uncoded.

From each input bit, one output bit is in each case generated by the coding with G12 or, respectively, G13. These appear successively at the output of the coder.

Thus, a serial input sequence of 255 input bits results in a serial sequence of 510 coded bits {C(0)... C(509)} at the output of the coder, which is defined by:

$C(2k) = u(k)$

$C(2k+1) = u(k)+u(k-2)+u(k-4)+u(k-5)+C(2k-1)+C(2k-3)+C(2k-5)+C(2k-9)$

for k = 0, 1, ..., 254; u(k) = 0 for k<0: C(k) = 0 for k<0

The bits at the output are thus coded alternately with G12 and G13.

The code is punctured so that the following 62 coded bits: {C(4*j+1) for j = 79, 80, ..., 127)} and the bits C(363), C(379), C(395), C(411), C(427), C(443), C(459), C(475), C(491), C(495), C(499), C(503) and C(507) are not transmitted.

As a result, there is a block of 448 coded and punctured bits, P(0)...P(447) which is appended to the bits of an in-band signaling in c.

$c(k+8) = P(k)$ for $k = 0, 1, ..., 447$.

CH1-FS:

A block of 215 bits {u(0)... u(214)} is coded at the rate 1/3, using the following polynomials:

$$G12 = 1$$

$$G13 = (1 + D^2 + D^4 + D^5) / (1 + D + D^2 + D^3 + D^5)$$

$$G14 = (1 + D^3 + D^4 + D^5) / (1 + D + D^2 + D^3 + D^5)$$

resulting in 645 coded bits, {C(0)...C(645)} defined by:

$$C(3k) = u(k)$$

$$C(3k+1) = u(k)+u(k-2)+u(k-4)+u(k-5)+C(3k-2)+C(3k-5)+C(3k-8)+C(3k-14)$$

$$C(3k+2) = u(k)+u(k-3)+u(k-4)+u(k-5)+C(3k-1)+C(3k-4)+C(3k-7)+C(3k-13)$$

for $k = 0, 1, ..., 214$; $u(k) = 0$ for $k<0$; $C(k) = 0$ for $k<0$

The code is punctured so that the following 197 coded bits:

{C(12*j+5), C(12*j+8), C(12*j+11) for j = 0, 1, ..., 25, {C(12*j+2), C(12*j+5), C(12*j+8), C(12*j+11)

for j = 26, 27, ..., 52}

and the bits C(2), C(610), C(622), C(628), C(634), C(637), C(638), C(640), C(641), C(643) and C(644) are not transmitted.

As a result, there is a block of 448 coded and punctured bits, P(0)...P(447) which is appended to the bits of an in-band signaling in c.

$c(k+8) = P(k)$ for $k = 0, 1, ..., 447$.

CH2-FS:

A block of a=171 bits {u(0)... u(170)} is coded at the rate 1/3, using the following polynomials:

$$G12 = 1$$

$$G15 = (1 + D + D^2 + D^3 + D^6) / (1 + D^2 + D^3 + D^5 + D^6)$$

$$G16 = (1 + D + D^4 + D^6) / (1 + D^2 + D^3 + D^5 + D^6)$$

resulting in 513 coded bits, {C(0)... C(512)} defined by:

C(3k) = u(k)

C(3k+1) = u(k)+u(k-1)+u(k-2)+u(k-3)+u(k-6)+C(3k-5)+ C(3k-8)+C(3k-14)+C(3k-17)

C(3k+2) = u(k)+u(k-1)+u(k-4)+u(k-6)+C(3k-4)+C(3k-7)+ C(3k-11)+C(3k-16)

for k = 0, 1, ..., 170; u(k) = 0 for k<0; C(k) = 0 for k<0

The code is punctured so that the following 65 coded bits: {C(21*j+20) for j = 0, 1, ..., 15 C(21*j+8) C(21*j+11) C(21*j+17) C(21*j+20) for j = 16, 17, ..., 23} and the bits C(1), C(2), C(4), C(5), C(8), C(326), C(332), C(488), C(497), C(499), C(502), C(505), C(506), C(508), C(509), C(511) and C(512) are not transmitted.

As a result, there is a block of 448 coded and punctured bits, P(0)...P(447) which is appended to the bits of an in-band signaling in c.

c(k+8) = P(k) for k = 0, 1, ..., 447.

The polynomials used in modes CH5-FS, CH6-FS, CH7-FS are:

$$G17 = (1 + D^2 + D^3 + D^4 + D^5 + D^6) / (1 + D^2 + D^3 + D^5 + D^6)$$

The significant values for modes (CH3-FS, CH4-FS, CH5-FS, CH6-FS, CH7-FS) are:

CH3-FS:

C(3k) = u(k)

C(3k+1) = u(k)+u(k-1)+u(k-2)+u(k-3)+u(k-6)+C(3k-5)+ C(3k-8)+C(3k-14)+C(3k-17)

C(3k+2) = u(k)+u(k-1)+u(k-4)+u(k-6)+C(3k-4)+C(3k-7)+ C(3k-11)+C(3k-16)

for k = 0, 1, ..., 159; u(k) = 0 for k<0; C(k) = 0 for k<0

Bits {C(18*j+2), C(21*j+8), C(21*j+11), C(21*j+17) for j = 20, 21, ..., 25} and C(353), C(359), C(470), C(473), C(475), C(476), C(478), C(479) are not transmitted.

CH4-FS:

C(4k) = u(k)

C(4k+1) = u(k)+u(k-1)+u(k-2)+u(k-3)+u(k-6)+C(4k-7)+ C(4k-11)+C(4k-19)+C(4k-23)

C(4k+2) = u(k)+u(k-1)+u(k-4)+u(k-6)+C(4k-6)+C(4k-10)+ C(4k-18)+C(4k-22)

C(4k+3) = u(k)+u(k-2)+u(k-3)+u(k-4)+u(k-5)+u(k-6)+ C(4k-5)+C(4k-9)+C(4k-17)+C(4k-21)

for k = 0, 1, ..., 145; u(k) = 0 for k<0; C(k) = 0 for k<0

Bits {C(32*j+7), C(32*j+15), C(32*j+23), C(32*j+27) C(32*j+31) for j = 0, 1, ..., 10 C(16*j+3) C(16*j+7) C(16*j+11) C(16*j+14) C(16*j+15) for j = 22, 23, ..., 35} and bits C(2), C(3), C(11), C(331), C(566), C(570), C(578), C(579), C(581), C(582) and C(583) are not transmitted.

CH5-FS:

C(4k) = u(k)

C(4k+1) = u(k)+u(k-1)+u(k-2)+u(k-3)+u(k-6)+C(4k-7)+ C(4k-11)+C(4k-19)+C(4k-23)

C(4k+2) = u(k)+u(k-1)+u(k-4)+u(k-6)+C(4k-6)+C(4k-10)+ C(4k-18)+C(4k-22)

C(4k+3) = u(k)+u(k-2)+u(k-3)+u(k-4)+u(k-5)+u(k-6)+ C(4k-5)+C(4k-9)+C(4k-17)+C(4k-21)

for k = 0, 1, ..., 129; u(k) = 0 for k<0; C(k) = 0 for k<0

Bits {C(32*j+11), C(32*j+23), C(32*j+31) for j = 0, 1, ..., 9

C(32*j+7), C(32*j+11), C(32*j+15), C(32*j+23), C(32*j+27), C(32*j+31) for j = 10, 11, ..., 15}

and bits C(499), C(510), C(514), C(515), C(518), C(519) are not transmitted.

CH6-FS:

C(4k) = u(k)

C(4k+1) = u(k)+u(k-1)+u(k-2)+u(k-3)+u(k-6)+C(4k-7)+ C(4k-11)+C(4k-19)+C(4k-23)

C(4k+2) = u(k)+u(k-1)+u(k-4)+u(k-6)+C(4k-6)+C(4k-10)+ C(4k-18)+C(4k-22)

C(4k+3) = u(k)+u(k-2)+u(k-3)+u(k-4)+u(k-5)+u(k-6)+ C(4k-5)+C(4k-9)+C(4k-17)+C(4k-21)

for k = 0, 1, ..., 114; u(k) = 0 for k<0; C(k) = 0 for k<0

Bits

{C(16*j+11) for j = 22, 23, ..., 28} and bits C(450), C(451), C(454), C(455), C(458) are not transmitted.

CH7-FS:

C(4k) = u(k)

C(4k+1) = u(k)+u(k-1)+u(k-2)+u(k-3)+u(k-6)+C(4k-7)+ C(4k-11)+C(4k-19)+C(4k-23)

C(4k+2) = u(k)+u(k-1)+u(k-4)+u(k-6)+C(4k-6)+C(4k-10)+ C(4k-18)+C(4k-22)

C(4k+3) = u(k)+u(k-2)+u(k-3)+u(k-4)+u(k-5)+u(k-6)+ C(4k-5)+C(4k-9)+C(4k-17)+C(4k-21)

for k = 0, 1, ..., 94; u(k) = 0 for k<0; C(k) = 0 for k<0

Bits

C(1), C(2), C(3), C(6), C(7), C(11), C(367), C(383), C(399), C(407), C(415), C(418), C(419), C(421), C(422), C(423), C(425), C(426), C(427) are removed. In this block of 409 coded and punctured bits, P(0)... P(408), 39 bits are repeated:

$P(409+k) = P(10+k*8)$ for $k = 0, 1, ..., 38$

Coding in half-rate mode (HR)

Coding of the bits of the in-band signaling:

id(0,1)	ic(0..3)
00	0000
01	1001
10	0111
11	1110

Distribution of the bits to classes:

Coding mode	Number of voice bits per block	Number of Class-1 bits per block	Number of Class-1a bits per block	Number of Class-1b bits per block	Number of Class-2 bits per block
CH8-HS	159	123	67	56	36
CH9-HS	148	120	61	59	28
CH10-HS	134	110	55	55	24
CH11-HS	118	102	55	47	16
CH12-HS	103	91	49	42	12
CH13-HS	95	83	39	44	12

The essential parameters for the coder and correspondingly for each decoder are specified as follows for the first coding step:

Coding mode	Number of Class 1 bits (K_{d1})	CRC-protected bits (K_{dts})	Number of tail bits (N_{tail})	Number of output bits after the first coding step ($K_u=K_{d1}+6+N_{tail}$)
CH8-HS	123	67	5	134
CH9-HS	120	61	5	131
CH10-HS	110	55	5	121
CH11-HS	102	55	5	113
CH12-HS	91	49	6	103
CH13-HS	83	39	6	95

The information on the parity and tail bits and on the reordering corresponding to the full-rate mode.

After the first coding step u(k), the following contents are defined for each coding mode:

CH8-HS: $u(k) = d(k)$ for $k = 0, 1, ..., 66$

$u(k) = p(k-67)$ for $k = 67, 68, ..., 72$

$u(k) = d(k-6)$ for $k = 73, 74, ..., 128$

$u(k) =$ to be specified for $k = 129, 130, ..., 133$

CH9-HS: $u(k) = d(k)$ for $k = 0, 1, ..., 60$

$u(k) = p(k-61)$ for $k = 61, 62, ..., 66$

$u(k) = d(k-6)$ for $k = 67, 68, ..., 125$

$u(k) =$ to be specified for $k = 126, 127, ..., 130$

CH10-HS: $u(k) = d(k)$ for $k = 0, 1, ..., 54$

$u(k) = p(k-55)$ for $k = 55, 56, ..., 60$

$u(k) = d(k-6)$ for $k = 61, 62, ..., 115$

$u(k) =$ to be specified for $k = 116, 117, ..., 120$

CH11-HS: $u(k) = d(k)$ for $k = 0, 1, ..., 54$

$u(k) = p(k-55)$ for $k = 55, 56, ..., 60$

$u(k) = d(k-6)$ for $k = 61, 62, ..., 107$

$u(k) =$ to be specified for $k = 108, 109, ..., 112$

CH12-HS: $u(k) = d(k)$ for $k = 0, 1, ..., 48$

$u(k) = p(k-49)$ for $k = 49, 50, ..., 54$

$u(k) = d(k-6)$ for $k = 55, 56, ..., 96$

$u(k) =$ to be specified for $k = 97, 98, ..., 102$

CH13-HS: $u(k) = d(k)$ for $k = 0, 1, ..., 38$

$u(k) = p(k-39)$ for $k = 39, 40, ..., 44$

$u(k) = d(k-6)$ for $k = 45, 46, ..., 88$

$u(k) =$ to be specified for $k = 89, 90, ..., 94$

Convolutional coder

The bits of the first coding step (u(k)) are coded with a recursive systematic convolutional code (see also Figure 4). The number of output bits after puncturing and repetition is 448 bits for all modes of the coding method.

Coding mode	Generator polynomials of convolutional code	Number of bits received in the coder	Coder rate	Number of bits output by the coder	Number of punctured bits
CH8-HS	G12, G13	134	1/2	268	80
CH9-HS	G12, G13	131	1/2	262	66
CH10-HS	G12, G13	121	1/2	242	42
CH11-HS	G12, G13	113	1/2	226	18
CH12-HS	G12, G15, G16	103	1/3	309	97
CH13-HS	G12, G15, G16	95	1/3	285	73

The coding modes are presented in sequence:

CH8-HS:

One block of 134 bits {u(0)...u(133)} each is coded at the rate of 1/2, using the following polynomials:

$$G12 = 1$$

$$G13 = (1 + D^2 + D^4 + D^5) / (1 + D + D^2 + D^3 + D^5)$$

resulting in 268 coded bits, {C(0)...C(267)}, defined by:

C(2k) = u(k)

C(2k+1) = u(k)+u(k-2)+u(k-4)+u(k-5)+C(2k-1)+C(2k-3)+C(2k-5)+C(2k-9)

for k = 0, 1, ..., 133; u(k) = 0 for k<0; C(k) = 0 for k<0

The code is punctured so that the following 80 coded bits:

{C(8*j+3), C(8*j+7) for j = 0, 1, ..., 21

C(8*j+3), C(8*j+5), C(8*j+7) for j = 22, 23, ..., 32)} and the bits C(1), C(265) and C(267) are not transmitted.

As a result, there is a block of 188 coded and punctured bits, P(0)...P(187) which is appended to the bits of an in-band signaling in c.

c(k+4) = P(k) for k = 0, 1, ..., 187.

Finally, 36 Class-2 bits are appended to c

c(192+k) = d(123+k) for k = 0, 1, ..., 35.

CH9-HS:

The 262 coded bits $\{C(0)...C(261)\}$

$C(2k) = u(k)$

$C(2k+1) = u(k)+u(k-2)+u(k-4)+u(k-5)+C(2k-1)+C(2k-3)+C(2k-5)+C(2k-9)$

for $k = 0, 1, ..., 130$; $u(k) = 0$ for $k<0$; $C(k) = 0$ for $k<0$

are punctured so that 66 coded bits:

$\{C(16*j+3), C(16*j+7), C(16*j+11)$ for $j = 0, 1, ..., 7$

$C(16*j+3), C(16*j+7), C(16*j+11), C(16*j+15)$ for $j = 8, 9, ..., 15)\}$ and the bits C(1),

C(221), C(229), C(237), C(245), C(249), C(253), C(257), C(259) and C(261) are not transmitted.

A block of 196 coded and punctured bits, P(0)...P(195) is appended to the bits of the in-band signaling in c:

$c(k+4) = P(k)$ for $k = 0, 1, ..., 195$.

Finally, 28 Class-2 bits are appended to c:

$c(200+k) = d(120+k)$ for $k = 0, 1, ..., 27$.

CH10-HS:

The 242 coded bits $\{C(0)...C(241)\}$:

$C(2k) = u(k)$

$C(2k+1) = u(k)+u(k-2)+u(k-4)+u(k-5)+C(2k-1)+C(2k-3)+C(2k-5)+C(2k-9)$

for $k = 0, 1, ..., 106$; $u(k) = 0$ for $k<0$; $C(k) = 0$ for $k<0$

are punctured so that 42 coded bits:

$\{C(8*j+3)$ for $j = 0, 1, ..., 21$

$C(8*j+3), C(8*j+7)$ for $j = 22, 23, ..., 29)\}$ and the bits C(1), C(233), C(237) and C(241) are not transmitted.

A block of 200 coded and punctured bits, P(0)...P(199) is appended to the bits of the in-band signaling in c:

c(k+4) = P(k) for k = 0, 1, ..., 199.

Finally, 24 Class-2 bits are appended to c:

c(204+k) = d(110+k) for k = 0, 1, ..., 23.

CH11-HS:

The 226 coded bits {C(0)...C(225)}:

C(2k) = u(k)

C(2k+1) = u(k)+u(k-2)+u(k-4)+u(k-5)+C(2k-1)+C(2k-3)+ C(2k-5)+C(2k-9)

for k = 0, 1, ..., 112; u(k) = 0 for k<0; C(k) = 0 for k<0

are punctured so that 18 coded bits:

{C(28*j+15) for j = 0, 1, ..., 7} and bits C(1), C(3), C(7), C(197), C(213), C(215), C(217), C(221), C(223) and C(225) are not transmitted.

A block of 208 coded and punctured bits, P(0)...P(207) is appended to the bits of the in-band signaling in c:

c(k+4) = P(k) for k = 0, 1, ..., 207.

Finally, 16 Class-2 bits are appended to c:

c(212+k) = d(96+k) for k = 0, 1, ..., 15.

CH12-HS:

The 309 coded bits {C(0)...C(308)}:

C(3k) = u(k)

C(3k+1) = u(k)+u(k-1)+u(k-2)+u(k-3)+u(k-6)+C(3k-5)+ C(3k-8)+C(3k-14)+C(3k-17)

C(3k+2) = u(k)+u(k-1)+u(k-4)+u(k-6)+C(3k-4)+C(3k-7)+ C(3k-11)+C(3k-16)

for k = 0, 1, ..., 102; u(k) = 0 for k<0; C(k) = 0 for k<0

are punctured so that 97 coded bits:

{C(12*j+5), C(12*j+8), C(12*j+11) for j = 0, 1, ..., 15 C(12*j+2), C(12*j+5), C(12*j+8), C(12*j+11) for j = 16, 17, ..., 24) and bits C(1), C(2), C(4), C(7), C(292), C(292), C(295), C(298), C(301), C(302), C(304), C(305), C(307) and C(308) are not transmitted.

A block of 212 coded and punctured bits, P(0)...P(211) is appended to the bits of the in-band signaling in c:

c(k+4) = P(k) for k = 0, 1, ..., 211.

Finally, 12 Class-2 bits are appended to c:

c(216+k) = d(91+k) for k = 0, 1, ..., 11.

CH13-HS:

The 285 coded bits {C(0)...C(284)}:

C(3k) = u(k)

C(3k+1) = u(k)+u(k-1)+u(k-2)+u(k-3)+u(k-6)+C(3k-5)+ C(3k-8)+C(3k-14)+C(3k-17)

C(3k+2) = u(k)+u(k-1)+u(k-4)+u(k-6)+C(3k-4)+C(3k-7)+ C(3k-11)+C(3k-16)

for k = 0, 1, ..., 94; u(k) = 0 for k<0; C(k) = 0 for k<0

are punctured so that 73 coded bits:

{C(12*j+5), C(12*j+11) for j = 0, 1, ..., 11 C(12*j+5), C(12*j+8), C(12*j+11) for j = 12, 13, ..., 22} and bits C(1), C(2), C(4), C(7), C(8), C(14), C(242), C(254), C(266), C(274), C(277), C(278), C(280), C(281), C(283) and C(284) are not transmitted.

A block of 212 coded and punctured bits, P(0)...P(211) is appended to the bits of the in-band signaling in c:

c(k+4) = P(k) for k = 0, 1, ..., 211.

Finally, 12 Class-2 bits are appended to c:

c(216+k) = d(91+k) for k = 0, 1, ..., 11.

The polynomials of the systematic recursive code (G13 to G17) in the AMR (see Figure 5) shown were used for two reasons:

- they have optimum characteristics for the puncturing~~,~~; i.e., the adaptation of the data rate to the transmission rate of the radio channel, and
- numerator or denominator polynomial are in each case also the polynomial used in the original AMR channel coding proposal (see Tdoc SMG 147/98). The necessary changes are thus minimum compared with the original proposal.

The polynomials used ~~hitherto~~ for voice, data and signaling information in the GSM system can also be used for the AMR channel coder with negligible restrictions in the performance. This can be done instead of the polynomials described above or as a complete alternative channel coding arrangement. The advantage lies in that the compatibility is extended further since in some cases older pre-existing hardware components in the channel decoder only allow the previous GSM polynomials to be used.

Figure 6 shows a base station BS in which, in the reception case, signals received via an antenna A are amplified in a receiver, filtered, converted to baseband and digitized.

This is followed by channel decoding (step 1), which can be done with decoding devices installed in existing base stations BS~~,~~; i.e., the circuit technology can remain unchanged. This is followed by post-processing (step 2) of the decoded data which is implemented as a program. This post-processing consists of convolutional coding at a rate of 1 with the denominator polynomial of the respective rate.

As a result, this post-processing is of little complexity and is performed, for example, by an additional program in a DSP (digital signal processor).

Referring, e.g. to the rate CH0-FS, this ~~means that~~ refers to the block with 255 bits at the output of the decoder ~~must be~~ being coded with the polynomial:

$$G(D) = (1 + D + D^2 + D^3 + D^5)$$

in order to obtain the 255 original bits. The number of data bits remains constant~~,~~; i.e., a current data bit at the input of this post-processing yields exactly one original bit with the aid of past input bits.

The coding and decoding methods described can be used both in base stations BS and in mobile stations MS.

Although the present invention has been described with reference to specific embodiments, those of skill in the art will recognize that changes may be made thereto without departing from the spirit and scope of the invention as set forth in the hereafter appended claims.

~~Abstract~~

ABSTRACT OF THE DISCLOSURE

~~Method, base station and subscriber station for channel coding in a GSM mobile radio system~~

A method, base station and subscriber station which ~~According to the invention, it is proposed to~~ use recursive systematic codes (RSC codes) for channel coding in GSM mobile radio systems. In contrast to previous conceptions, these RSC codes ~~can~~ also can be used on the basis of the hardware installed in existing GSM mobile radio systems. The RSC codes can be introduced during the introduction of an adaptive multirate coder ~~(AMR)~~.

~~Figure 5~~

~~Key to figures~~

~~Figure 1:~~
~~Stand der Technik = Prior art~~

~~Figure 2:~~
~~Nonsystematic nonrecursive code with memory 4 and rate 1/2 analogously to GSM/TCHFS~~

~~Figure 3:~~
~~Identical recursive systematic convolutional code with memory 4 and rate 1/2~~

~~Figure 4:~~
~~1 In-band data~~
~~2 Voice frames~~
~~3 Sorting~~
~~4 Class 2~~
~~5 Block code~~
~~6 Convolutional code~~
~~7 or~~
~~8 Reordering and distribution~~
~~9 Diagonal interleaving~~
~~in: 4 blocks~~
~~out: block pairs~~
~~10 Encryption~~

~~Figure 5:~~
~~Polynomials used in different channels in the GSM mobile radio system~~
~~1 User data channel, adaptive multirate coding, full rate~~
~~2 User data channel, adaptive multirate coding, half rate~~

~~Figure 6:~~
~~1 Receiver~~
~~2 Channel decoder~~
~~3 Post-processing~~

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INVENTION: METHOD, BASE STATION AND SUBSCRIBER STATION FOR CHANNEL CODING IN A GSM MOBILE RADIO SYSTEM

Assistant Commissioner for Patents,
Washington, D.C. 20231

SUBMISSION OF DRAWINGS

Applicants herewith submit five sheets (Figs. 1-6) of drawings for the above-referenced PCT application.

Respectfully submitted,

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Description

Method, base station and subscriber station for channel coding in a GSM mobile radio system

The invention relates to a method, base station and subscriber station for channel coding in a GSM mobile radio system.

The GSM (global system for mobile communications) mobile radio system is installed in more than 100 networks and for more than 100 million subscribers worldwide. In the GSM mobile radio system, data (for example voice or data within data services such as SMS or GPRS) are transmitted via a radio interface with the aid of electromagnetic waves. The radio interface relates to a connection between a base station and subscriber stations where the subscriber stations can be mobile stations or stationary radio stations. The electromagnetic waves are radiated in this case with carrier frequencies which are within the frequency bands of 900, 1800, 1900 MHz in the GSM mobile radio system.

In mobile radio systems, channel coding is required for transmitting the data via the radio interface. This channel coding differs for different services, e.g. 14.4 kbps data, FR (full-rate) voice, HR (half-rate) voice. The channel coding and the complementary channel decoding at the receiving end have the aim here of achieving the lowest possible bit error rate (BER).

Hitherto, only nonsystematic nonrecursive convolutional codes (NSC - nonsystematic convolutional codes) have been used for channel coding in the GSM mobile radio system (and other comparable systems). In these codes, a coded bit is generated from a weighted sum of the current and past information bits by convolutional coding. At a coding rate of ½, e.g. 2 coded bits, which in each case come from

a differently weighted sum, are thus generated from one information bit (see Figure 2). The weights in this sum, and thus the generation of the coded bits, are determined by the so-called generator polynomials. Thus, e.g., the polynomial $1 + D^3 + D^4$ determines that a coded bit is produced from the sum (XOR combination) of the current, the third last and the fourth last information bit.

The bits coded during the channel coding are transmitted via the radio interface and channel-decoded at the receiving end. A frequently used decoding algorithm is the so-called Viterbi algorithm. Since the decoding process remains the same and is also computationally intensive, hardware chips (application-specific integrated circuits (ASICs)) are used for this purpose, especially in base stations. As a rule, these ASICs can only process a certain decoding scheme, only for nonrecursive currents in the case of GSM.

In the case of the introduction of a new voice coding message for GSM mobile radio systems, the methods hitherto proposed for the channel coding, see ETSI SMG11; Tdoc SMG11 205/98, 159/98 and 147/98, 9.28.98, are exclusively based on nonrecursive codes in order to ensure compatibility with the existing hardware which is used in millions. In spite of the involvement of many manufacturers in the development process, see Tdoc SMG11 205/98, 159/98 and 147/98, of 9.28.98, other types of code have been considered to be unusable.

The invention is based on the object of specifying a method for channel coding and corresponding devices which produce better transmission quality. This object is achieved by the method having the features of claim 1 and the devices having the features of claims 10 and 11, respectively.

According to the invention, it is proposed to use recursive systematic codes (RSC codes) for the channel coding. These

differ from the NSC codes in that, e.g. at a rate of ½, the first "coded" bit corresponds to the current information bit (systematic) and the second coded bit is produced from the current and past information bits and past coded bits (recursive). Thus, codes which are fed back are used, making use of the fact that recursive systematic codes have distinctly better code characteristics, and thus also better characteristics with respect to the error correction, especially at high bit error rates.

The RSC codes, known from, among others, E. Offer, "Decodierung mit Qualitätsinformation bei verketteten Codiersystemen" [Decoding with quality information in concordinated coding systems], progress reports, VDI-Verlag, Series 10, Vol. 443, Düsseldorf 1996, p. 21 ff and p. 119 ff, have previously not been used since they result in changes in the decoding process and are thus not hardware-compatible. An introduction of RSC codes in the channel coding did not appear possible since the installed base stations had to be retrofitted. This is not the case, in fact, since the hardware structure can be retained both at the transmitting end and at the receiving end and, nevertheless, RSC codes can be introduced for channel decoding in the GSM mobile radio system.

It is proposed to perform post-processing on the basis of the denominator polynomial with parts of the recursive systematic code after channel decoding at the receiving end. According to an advantageous further development of the invention, the decoding process is performed as previously with decoding of a NSC code, namely the one which is identical to the nonrecursive component - the numerator polynomial - of the new RSC code. After this hardware-compatible decoding, post-processing follows in which the bits obtained by this means are again coded with the denominator polynomial. This post-processing is advantageously performed via

programming means, that is to say in software, which can be more easily loaded into existing stations later.

The coding of the post-processing is not computationally expensive and can be performed as an additional step in every base station. This recoding provides the exact bits of the data sequence of the transmitting end.

A recursive decoding which is not possible with previously installed hardware can be replaced by decoding into two nonrecursive successive individual steps. The first step is decoding using the numerator polynomial of the recursive code and the second step is a coding using the denominator polynomial of the recursive code. This makes it possible to reproduce, if necessary, any systematic recursive codes using hardware which has already been installed. The first step corresponds to the previous decoding and the second step is the post-processing.

The polynomials of identical RSC and NSC codes will be explained briefly by means of Figures 2 and 3. In a typical NSC code (such as, e.g. GSM/TCHFS).

The generator polynomials there are:

Polynomials of the NSC codes: $G_1 = 1 + D^3 + D^4$

$G_2 = 1 + D + D^3 + D^4$

An identical RSC code is generated by dividing, e.g. by G_1:

$G_1 = 1$

Polynomials of the RSC code:

$$G_2 = \frac{1 + D + D^3 + D^4}{1 + D^3 + D^4} \ldots.$$

These RSC codes have the advantage that lower bit error rates are possible in the case of core channels (up to a bit error rate of 10^{-4}) since the channel error rate is not exceeded due to the uncoded bits (systematic component). In contrast, the bit error rate of coded bits can also be greater than the channel error rate under very poor channel conditions.

According to an advantageous development of the invention, a priori knowledge is obtained from a previous detection at the receiving end and this a priori knowledge is used in a subsequent channel decoding. During the transmission of coded voice, a number of voice parameters, and thus bits, change only rarely or it is also possible to make predictions of the probable current value from the value these parameters in the past. If then the received current value distinctly deviates from the predicted value, there is a high probability of a transmission error and, for example, the received value can be replaced by the predicted value.

This previous knowledge (a priori knowledge) is introduced in the channel decoder and has previously been impossible in most cases since the decoding algorithm had to be modified due to the use of non-systematic codes. As a rule, the modification was, in turn, not hardware-compatible. If RSC codes are used, this a priori knowledge can be introduced quite simply before the decoding process since some of the received bits are uncoded. The decoding process itself does not need to be modified.

As already explained, some of the received bits are uncoded information bits. If the channel conditions are good, i.e. no transmission errors are to be expected, channel decoding can be omitted and only the information bits are used. The transmission quality can then be determined as early as before the channel decoder by advantageously evaluating information from a channel estimator. After that, a decision is made as to whether decoding is necessary or not. In subscriber stations in which the energy consumption is an essential quality criterion, an essential advantage is that the channel decoder can be switched off. This saves power. In addition, the hardware for channel decoding can be omitted altogether in applications -

e.g. SMS applications for linking in telemetry services etc. -

in which a high transmission quality is always expected.

Due to a nonrecursive decoding followed by coding, it becomes possible to use RSC codes with the advantages described above in existing GSM mobile radio systems on existing hardware.

An exemplary embodiment of the invention is explained in greater detail on the basis of the network structure of the known GSM mobile radio system according to Figure 1 and referring to the codes according to Figures 2 and 3.

Figure 4 shows a flow chart of the coding,

Figure 5 shows polynomials used in the coding and decoding, and

Figure 6 shows a flow chart of the decoding.

The GSM mobile radio system shown in Figure 1 consists of a multiplicity of mobile switching centers MSC which are networked together and, respectively, establish access to a landline network PSTN. These mobile switching centers MSC are also connected to in each case at least one base station controller BSC for controlling base stations BS. Each of these base station controllers BSC, in turn, provides for a connection to at least one base station BS. An operation and maintenance center OMC implements control and maintenance functions for the mobile radio system or for parts thereof, respectively.

A base station BS can set up a connection to subscriber stations, e.g. mobile stations MS or other mobile and stationary terminals via a radio interface. Each base station BS forms at least one radio cell. Figure 1 shows connections for transmitting user information between a base station BS and mobile stations MS.

In the coding methods shown, voice information is transmitted as user information. The bits of the voice information are divided into three classes with respect to the weighting (Class 1a, 1b and 2) in accordance with their sensitivity to errors. The most important bits (Class 1a are additionally protected by a cyclic redundancy check (CRC) error protection coding. The bits of Classes 1a and 1b are convolutionally coded and punctured. In the AMR, the interleaving of the data after the coding is performed in accordance with the interleaving arrangements previously introduced for FR and HR.

Altogether, 14 coding methods are presented in conjunction with the AMR coder, from which a selection must be made in accordance with the transmission conditions. Of these, eight coding modes can be used in full-rate mode and six coding modes can be used in half-rate mode.

Trans-mission mode	**Channel coding mode**	**Source encoding bit rate, voice**	**Net bit rate, in-band signaling**	**Channel coding bit rate, voice**	**Channel coding bit rate, in-band**
TCH/FR	CH0-FS	12.20 kbit/s (GSM EFR)	0.10 bit/s	10.20 kbit/s	0.30 kbit/s
	CH1-FS	10.20 kbit/s	0.10 bit/s	12.20 kbit/s	0.30 kbit/s
	CH2-FS	7.95 kbit/s	0.10 bit/s	14.45 kbit/s	0.30 kbit/s
	CH3-FS	7.40 kbit/s (IS-641)	0.10 bit/s	15.00 kbit/s	0.30 kbit/s
	CH4-FS	6.70 kbit/s	0.10 bit/s	15.70 kbit/s	0.30 kbit/s
	CH5-FS	5.90 kbit/s	0.10 bit/s	16.50 kbit/s	0.30 kbit/s
	CH6-FS	5.15 kbit/s	0.10 bit/s	17.25 kbit/s	0.30 kbit/s
	CH7-FS	4.75 kbit/s	0.10 bit/s	17.65 kbit/s	0.30 kbit/s
TCH/HR	CH8-HS	7.95 kbit/s	0.10 bit/s	3.25 kbit/s	0.10 kbit/s
	CH9-HS	7.40 kbit/s (IS-41)	0.10 bit/s	3.80 kbit/s	0.10 kbit/s
	CH10-HS	6.70 kbit/s	0.10 bit/s	4.50 kbit/s	0.10 kbit/s
	CH11-HS	5.90 kbit/s	0.10 bit/s	5.30 kbit/s	0.10 kbit/s
	CH12-HS	5.15 kbit/s	0.10 bit/s	6.05 kbit/s	0.10 kbit/s
	CH13-HS	4.75 kbit/s	0.10 bit/s	6.45 kbit/s	0.10 kbit/s

An in-band signaling with 2 bits net (4 or, respectively, 8 bits gross after coding) per frame (20 ms) is used

for signaling the coding mode or for signaling the transmission quality in alternating frames. The two bits can be used for signaling four coding modes. These coding modes, which can be switched between by means of the in-band signaling, must be previously selected.

The following order of steps to be performed applies to all modes:

1. Information of the in-band signaling is coded with a block code,
2. The user information is sorted in accordance with their significance (class),
3. The ordered bits of the user information are coded with a systematic block code (CRC), generating words with voice and parity bits,
4. These coded bits and the rest of the Class 1 bits are convolutionally coded,
5. The coded bits are punctured in order to obtain the desired bit rate,
6. Unprotected bits are inserted into the frame with punctured data (only for half-rate mode),
7. The bits are reordered and the coded and in-band bits are interleaved, also inserting a so-called stealing flag.

The designations used in the [lacuna] have the following significance:

k, j	Numbering of the bits in data block or burst
K_x	Number of bits in a block, x specifies data type
n	Numbering of the output data blocks
N	A selected data block
B	Numbering of bursts or blocks
s(k)	Voice information before sorting, $k=1...K_s$ (interface 0 in Figure 4)
d(k)	Voice information before channel coding, $k=1...K_d-1$ (interface 1 in Figure 4)
id(k)	Bits of the in-band signaling, k=0,1

ic(k) Coded bits of the in-band signaling, k=0...3 (HR), 7 (FR)

u(k) Data after the first coding step, $k=0,1,...K_u-1$ (block coding, CRC coding) (interface 2 in Figure 4)

c(n,k), c(k) Data after the second coding step, $k=0,1..K_c-1$, n=0,1..N,N+1 (convolutional coding), (interface 3 in Figure 4)

i(B,k) Interleaved data, $k=0,1..K_j-1$, $B=B_0$, B_0+1, ..

e(B,k) Bits of a burst, k=0,1,114,115; $B=B_0$, B_0+1, .. (interface 4 in Figure 4)

Coding in full-rate mode (FR)

Coding of the bits of the in-band signaling:

id(0,1)	**ic(0..7)**
00	00000000
01	10111010
10	01011101
11	11100111

Distribution of the bits to classes:

Coding mode	**Number of voice bits per block**	**Number of Class-1 bits per block**	**Number of Class-1a bits per block**	**Number of Class-1b bits per block**
CH0-FS	244	244	81	163
CH1-FS	204	204	65	139
CH2-FS	159	159	75	84
CH3-FS	148	148	61	87
CH4-FS	134	134	55	79
CH5-FS	118	118	55	63
CH6-FS	103	103	49	54
CH7-FS	95	95	39	56

There are no class 2 bits.

The essential parameters for the coder and correspondingly for each decoder are specified as follows for the first coding step:

Coding mode	**Coded voice bits (K_d)**	**CRC-protected bits (K_{d1a})**	**Number of tail bits (N_{tail})**	**Number of bits after the first coding step ($K_s=K_d+6+N_{tail}$)**
CH0-FS	244	81	5	255
CH1-FS	204	65	5	215
CH2-FS	159	75	6	171
CH3-FS	148	61	6	160
CH4-FS	134	55	6	146
CH5-FS	118	55	6	130
CH6-FS	103	49	6	115
CH7-FS	95	39	6	107

a) Parity bits:

A 6-bit CRC (cyclic redundancy check) is used for error detection. These 6 parity bits are generated by using the following cyclic generator polynomial: $g(D) = D^6 + D^5 + D^3 + D^2 + D^1 + 1$ for the first K_{d1a} bits of Class 1, K_{d1a} specifying the number of bits of Class 1a according to the above table. The coding with the cyclic code is performed in systematic manner: in GF(2), the polynomials:

$d(0)D(K_{d1a}+5) + d(1)D(K_{d1a}+4) + \ldots + d(K_{d1a}-1)D^{(6)} + p(0)D^{(5)} + \ldots + p(4)D + p(5)$

where p(0), p(1) ... p(5) are the parity bits which, divided by g(D), give "0".

b) Tailing bits and reordering

The information bits and parity bits are brought together and so-called tail bits are appended:

$u(k) = d(k)$ for $k = 0, 1, \ldots, K_{d1a}-1$

$u(k) = p(k-K_{d1a})$ for $k = K_{d1a}, K_{d1a}+1, \ldots, K_{d1a}+5$

$u(k) = d(k-6)$ for $k = K_{d1a}+6, K_{d1a}+7, \ldots, K_d+5$

$u(k)$ = dependent on coding mode

Thus, the following contents are defined for each coding mode after the first coding step u(k):

CH0-FS: $u(k) = d(k)$ for $k = 0, 1, \ldots, 80$

$u(k) = p(k-81)$ for $k = 81, 82, \ldots, 86$

$u(k) = d(k-6)$ for $k = 87, 88, \ldots, 249$

$u(k)$ = to be specified for $k = 250, 251, \ldots, 254$

CH1-FS: u(k) = d(k) for k = 0, 1, ..., 64
u(k) = p(k-65) for k = 65, 66, ..., 70
u(k) = d(k-6) for k = 71, 72, ..., 209
u(k) = to be specified for k = 210, 211, ..., 214

CH2-FS: u(k) = d(k) for k = 0, 1, ..., 74
u(k) = p(k-75) for k = 75, 76, ..., 80
u(k) = d(k-6) for k = 81, 82, ..., 164
u(k) = to be specified for k = 165, 166, ..., 170

CH3-FS: u(k) = d(k) for k = 0, 1, ..., 60
u(k) = p(k-61) for k = 61, 62, ..., 66
u(k) = d(k-6) for k = 67, 68, ..., 153
u(k) = to be specified for k = 154, 155, ..., 159

CH4-FS: u(k) = d(k) for k = 0, 1, ..., 54
u(k) = p(k-55) for k = 55, 56, ..., 60
u(k) = d(k-6) for k = 61, 62, ..., 139
u(k) = to be specified for k = 140, 141, ..., 145

CH5-FS: u(k) = d(k) for k = 0, 1, ..., 54
u(k) = p(k-55) for k = 55, 56, ..., 60
u(k) = d(k-6) for k = 61, 62, ..., 123
u(k) = to be specified for k = 124, 125, ..., 129

CH6-FS: u(k) = d(k) for k = 0, 1, ..., 48
u(k) = p(k-49) for k = 49, 50, ..., 54
u(k) = d(k-6) for k = 55, 56, ..., 108
u(k) = to be specified for k = 109, 110, ..., 114

CH7-FS: u(k) = d(k) for k = 0, 1, ..., 38
u(k) = p(k-39) for k = 39, 40, ..., 44
u(k) = d(k-6) for k = 45, 46, ..., 100
u(k) = to be specified for k = 101, 102, ..., 106

Convolutional coder

The bits of the first coding step (u(k)) are coded with a recursive systematic convolutional code (see also Figure 4). The number of output bits after puncturing and repetition is 448 bits for all modes of the coding method.

Coding mode	Generator polynomials of convolutional code	Coder rate	Number of bits received in the coder	Number of bits output by the coder	Number of punctured bits	Number of repeated bits
CH0-FS	G12,G13	1/2	255	510	62	0
CH1-FS	G12,G13,G14	1/3	215	645	197	0
CH2-FS	G12,G15,G16	1/3	171	513	65	0
CH3-FS	G12,G15,G16	1/3	160	480	32	0
CH4-FS	G12,G15,G16	1/4	146	584	136	0
CH5-FS	G12,G15,G16,G17	1/4	130	520	72	0
CH6-FS	G12,G15,G16,G17	1/4	115	460	12	0
CH7-FS	G12,G15,G16,G17	1/4	107	428	19	39

Further details on coding/decoding using recursive codes were given in C. Berrou, A. Glavieux, "Near optimum error-correction coding and decoding: turbo codes" - "Reflections on the prize paper", IEEE Inf. Theory Soc. Newsletter, vol. 48, No. 2, June 1998 and C. Berrou and A. Glavieux: "Near optimum error-correcting coding and decoding: turbo codes", IEEE Trans. on Comm., vol. 44, pp. 1261-1271, October 1996.

The coding modes are presented in sequence:

CH0-FS:

A block of 255 bits {u(0)... u(254)} is coded at the rate 1/2, using the following polynomials:

$$G12 = 1$$

$$G13 = (1 + D^2 + D^4 + D^5) / (1 + D + D^2 + D^3 + D^5)$$

The coding with G12=1 means that the input bit is only multiplied by 1, i.e. is transmitted uncoded.

From each input bit, one output bit is in each case generated by the coding with G12 or, respectively, G13. These appear successively at the output of the coder.

Thus, a serial input sequence of 255 input bits results in a serial sequence of 510 coded bits {C(0)... C(509)} at the output of the coder, which is defined by:

$$C(2k) = u(k)$$
$$C(2k+1) = u(k)+u(k-2)+u(k-4)+u(k-5)+C(2k-1)+C(2k-3)+C(2k-5)+C(2k-9)$$

for k = 0, 1, ..., 254; u(k) = 0 for k<0: C(k) = 0 for k<0
The bits at the output are thus coded alternately with G12 and G13.

The code is punctured so that the following 62 coded bits: {C(4*j+1) for j = 79, 80, ..., 127)} and the bits C(363), C(379), C(395), C(411), C(427), C(443), C(459), C(475), C(491), C(495), C(499), C(503) and C(507) are not transmitted.

As a result, there is a block of 448 coded and punctured bits, P(0)...P(447) which is appended to the bits of an in-band signaling in c.

$$c(k+8) = P(k) \quad \text{for } k = 0, 1, ..., 447.$$

CH1-FS:

A block of 215 bits {u(0)... u(214)} is coded at the rate 1/3, using the following polynomials:

$$G12 = 1$$
$$G13 = (1 + D^2 + D^4 + D^5) / (1 + D + D^2 + D^3 + D^5)$$
$$G14 = (1 + D^3 + D^4 + D^5) / (1 + D + D^2 + D^3 + D^5)$$

resulting in 645 coded bits, {C(0)...C(645)} defined by:

$$C(3k) = u(k)$$
$$C(3k+1) = u(k)+u(k-2)+u(k-4)+u(k-5)+C(3k-2)+$$

C(3k-5)+C(3k-8)+C(3k-14)

C(3k+2) = u(k)+u(k-3)+u(k-4)+u(k-5)+C(3k-1)+ C(3k-4)+C(3k-7)+C(3k-13)

for k = 0, 1, ..., 214; u(k) = 0 for k<0; C(k) = 0 for k<0

The code is punctured so that the following 197 coded bits:

{C(12*j+5), C(12*j+8), C(12*j+11) for j = 0, 1, ..., 25,

{C(12*j+2), C(12*j+5), C(12*j+8), C(12*j+11) for j = 26, 27, ..., 52}

and the bits C(2), C(610), C(622), C(628), C(634), C(637), C(638), C(640), C(641), C(643) and C(644) are not transmitted.

As a result, there is a block of 448 coded and punctured bits, P(0)...P(447) which is appended to the bits of an in-band signaling in c.

c(k+8) = P(k) for k = 0, 1, ..., 447.

CH2-FS:

A block of a=171 bits {u(0)... u(170)} is coded at the rate 1/3, using the following polynomials:

G12 = 1

G15 = $(1 + D + D^2 + D^3 + D^6) / (1 + D^2 + D^3 + D^5 + D^6)$

G16 = $(1 + D + D^4 + D^6) / (1 + D^2 + D^3 + D^5 + D^6)$

resulting in 513 coded bits, {C(0)... C(512)} defined by:

C(3k) = u(k)

C(3k+1) = u(k)+u(k-1)+u(k-2)+u(k-3)+u(k-6)+C(3k-5)+ C(3k-8)+C(3k-14)+C(3k-17)

C(3k+2) = u(k)+u(k-1)+u(k-4)+u(k-6)+C(3k-4)+C(3k-7)+ C(3k-11)+C(3k-16)

for k = 0, 1, ..., 170; u(k) = 0 for k<0; C(k) = 0 for k<0

The code is punctured so that the following 65 coded bits:

{C(21*j+20) for j = 0, 1, ..., 15

C(21*j+8) C(21*j+11) C(21*j+17) C(21*j+20) for j = 16, 17, ..., 23} and the bits C(1), C(2), C(4), C(5), C(8), C(326), C(332), C(488), C(497), C(499), C(502), C(505), C(506), C(508), C(509), C(511) and C(512) are not transmitted.

As a result, there is a block of 448 coded and punctured bits, P(0)...P(447) which is appended to the bits of an in-band signaling in c.

c(k+8) = P(k) for k = 0, 1, ..., 447.

The polynomials used in modes CH5-FS, CH6-FS, CH7-FS are:

$$G17 = (1 + D^2 + D^3 + D^4 + D^5 + D^6) / (1 + D^2 + D^3 + D^5 + D^6)$$

The significant values for modes (CH3-FS, CH4-FS, CH5-FS, CH6-FS, CH7-FS) are:

CH3-FS:

C(3k) = u(k)

C(3k+1) = u(k)+u(k-1)+u(k-2)+u(k-3)+u(k-6)+C(3k-5)+C(3k-8)+C(3k-14)+C(3k-17)

C(3k+2) = u(k)+u(k-1)+u(k-4)+u(k-6)+C(3k-4)+C(3k-7)+C(3k-11)+C(3k-16)

for k = 0, 1, ..., 159; u(k) = 0 for k<0; C(k) = 0 for k<0

Bits {C(18*j+2), C(21*j+8), C(21*j+11), C(21*j+17) for j = 20, 21, ..., 25} and C(353), C(359), C(470), C(473), C(475), C(476), C(478), C(479) are not transmitted.

CH4-FS:

C(4k) = u(k)

C(4k+1) = u(k)+u(k-1)+u(k-2)+u(k-3)+u(k-6)+C(4k-7)+C(4k-11)+C(4k-19)+C(4k-23)

C(4k+2) = u(k)+u(k-1)+u(k-4)+u(k-6)+C(4k-6)+C(4k-10)+C(4k-18)+C(4k-22)

C(4k+3) = u(k)+u(k-2)+u(k-3)+u(k-4)+u(k-5)+u(k-6)+

C(4k-5)+C(4k-9)+C(4k-17)+C(4k-21)

for k = 0, 1, ..., 145; u(k) = 0 for k<0; C(k) = 0 for k<0

Bits {C(32*j+7), C(32*j+15), C(32*j+23), C(32*j+27)

C(32*j+31) for j = 0, 1, ..., 10

C(16*j+3) C(16*j+7) C(16*j+11) C(16*j+14) C(16*j+15) for j = 22, 23, ..., 35} and bits C(2), C(3), C(11), C(331), C(566), C(570), C(578), C(579), C(581), C(582) and C(583) are not transmitted.

CH5-FS:

C(4k) = u(k)

C(4k+1) = u(k)+u(k-1)+u(k-2)+u(k-3)+u(k-6)+C(4k-7)+C(4k-11)+C(4k-19)+C(4k-23)

C(4k+2) = u(k)+u(k-1)+u(k-4)+u(k-6)+C(4k-6)+C(4k-10)+C(4k-18)+C(4k-22)

C(4k+3) = u(k)+u(k-2)+u(k-3)+u(k-4)+u(k-5)+u(k-6)+C(4k-5)+C(4k-9)+C(4k-17)+C(4k-21)

for k = 0, 1, ..., 129; u(k) = 0 for k<0; C(k) = 0 for k<0

Bits

{C(32*j+11), C(32*j+23), C(32*j+31) for j = 0, 1, ..., 9

C(32*j+7), C(32*j+11), C(32*j+15), C(32*j+23), C(32*j+27), C(32*j+31) for j = 10, 11, ..., 15}

and bits C(499), C(510), C(514), C(515), C(518), C(519) are not transmitted.

CH6-FS:

C(4k) = u(k)

C(4k+1) = u(k)+u(k-1)+u(k-2)+u(k-3)+u(k-6)+C(4k-7)+C(4k-11)+C(4k-19)+C(4k-23)

C(4k+2) = u(k)+u(k-1)+u(k-4)+u(k-6)+C(4k-6)+C(4k-10)+C(4k-18)+C(4k-22)

C(4k+3) = u(k)+u(k-2)+u(k-3)+u(k-4)+u(k-5)+u(k-6)+C(4k-5)+C(4k-9)+C(4k-17)+C(4k-21)

for k = 0, 1, ..., 114; u(k) = 0 for k<0; C(k) = 0 for k<0

Bits
{C(16*j+11) for j = 22, 23, ..., 28} and bits C(450), C(451), C(454), C(455), C(458) are not transmitted.

CH7-FS:

C(4k) = u(k)

C(4k+1) = u(k)+u(k-1)+u(k-2)+u(k-3)+u(k-6)+C(4k-7)+ C(4k-11)+C(4k-19)+C(4k-23)

C(4k+2) = u(k)+u(k-1)+u(k-4)+u(k-6)+C(4k-6)+C(4k-10)+ C(4k-18)+C(4k-22)

C(4k+3) = u(k)+u(k-2)+u(k-3)+u(k-4)+u(k-5)+u(k-6)+ C(4k-5)+C(4k-9)+C(4k-17)+C(4k-21)

for k = 0, 1, ..., 94; u(k) = 0 for k<0; C(k) = 0 for k<0

Bits
C(1), C(2), C(3), C(6), C(7), C(11), C(367), C(383), C(399), C(407), C(415), C(418), C(419), C(421), C(422), C(423), C(425), C(426), C(427) are removed. In this block of 409 coded and punctured bits, P(0)... P(408), 39 bits are repeated:

P(409+k) = P(10+k*8) for k = 0, 1, ..., 38

Coding in half-rate mode (HR)

Coding of the bits of the in-band signaling:

id(0,1)	**ic(0..3)**
00	0000
01	1001
10	0111
11	1110

Distribution of the bits to classes:

Coding mode	**Number of voice bits per block**	**Number of Class-1 bits per block**	**Number of Class-1a bits per block**	**Number of Class-1b bits per block**	**Number of Class-2 bits per block**
CH8-HS	159	123	67	56	36
CH9-HS	148	120	61	59	28
CH10-HS	134	110	55	55	24
CH11-HS	118	102	55	47	16
CH12-HS	103	91	49	42	12
CH13-HS	95	83	39	44	12

The essential parameters for the coder and correspondingly for each decoder are specified as follows for the first coding step:

Coding mode	**Number of Class 1 bits (K_{d1})**	**CRC-protected bits (K_{dts})**	**Number of tail bits (N_{tail})**	**Number of output bits after the first coding step ($K_u=K_{d1}+6+N_{tail}$)**
CH8-HS	123	67	5	134
CH9-HS	120	61	5	131
CH10-HS	110	55	5	121
CH11-HS	102	55	5	113
CH12-HS	91	49	6	103
CH13-HS	83	39	6	95

The information on the parity and tail bits and on the reordering corresponding to the full-rate mode.

After the first coding step u(k), the following contents are defined for each coding mode:

CH8-HS:
$u(k) = d(k)$ for $k = 0, 1, \ldots, 66$
$u(k) = p(k-67)$ for $k = 67, 68, \ldots, 72$
$u(k) = d(k-6)$ for $k = 73, 74, \ldots, 128$
$u(k)$ = to be specified for $k = 129, 130, \ldots, 133$

CH9-HS:
$u(k) = d(k)$ for $k = 0, 1, \ldots, 60$
$u(k) = p(k-61)$ for $k = 61, 62, \ldots, 66$
$u(k) = d(k-6)$ for $k = 67, 68, \ldots, 125$
$u(k)$ = to be specified for $k = 126, 127, \ldots, 130$

CH10-HS:
$u(k) = d(k)$ for $k = 0, 1, \ldots, 54$
$u(k) = p(k-55)$ for $k = 55, 56, \ldots, 60$
$u(k) = d(k-6)$ for $k = 61, 62, \ldots, 115$
$u(k)$ = to be specified for $k = 116, 117, \ldots, 120$

CH11-HS:
$u(k) = d(k)$ for $k = 0, 1, \ldots, 54$
$u(k) = p(k-55)$ for $k = 55, 56, \ldots, 60$
$u(k) = d(k-6)$ for $k = 61, 62, \ldots, 107$

u(k) = to be specified for k = 108, 109, ..., 112

CH12-HS: u(k) = d(k) for k = 0, 1, ..., 48
u(k) = p(k-49) for k = 49, 50, ..., 54
u(k) = d(k-6) for k = 55, 56, ..., 96
u(k) = to be specified for k = 97, 98, ..., 102

CH13-HS: u(k) = d(k) for k = 0, 1, ..., 38
u(k) = p(k-39) for k = 39, 40, ..., 44
u(k) = d(k-6) for k = 45, 46, ..., 88
u(k) = to be specified for k = 89, 90, ..., 94

Convolutional coder

The bits of the first coding step (u(k)) are coded with a recursive systematic convolutional code (see also Figure 4). The number of output bits after puncturing and repetition is 448 bits for all modes of the coding method.

Coding mode	Generator polynomials of convolutional code	Number of bits received in the coder	Coder rate	Number of bits output by the coder	Number of punctured bits
CH8-HS	G12, G13	134	1/2	268	80
CH9-HS	G12, G13	131	1/2	262	66
CH10-HS	G12, G13	121	1/2	242	42
CH11-HS	G12, G13	113	1/2	226	18
CH12-HS	G12, G15, G16	103	1/3	309	97
CH13-HS	G12, G15, G16	95	1/3	285	73

The coding modes are presented in sequence:

CH8-HS:

One block of 134 bits {u(0)...u(133)} each is coded at the rate of 1/2, using the following polynomials:

G12 = 1

$$G13 = (1 + D^2 + D^4 + D^5) / (1 + D + D^2 + D^3 + D^5)$$

resulting in 268 coded bits, {C(0)...C(267)}, defined by:

C(2k) = u(k)

C(2k+1) = u(k)+u(k-2)+u(k-4)+u(k-5)+C(2k-1)+C(2k-3)+ C(2k-5)+C(2k-9)

for k = 0, 1, ..., 133; u(k) = 0 for k<0; C(k) = 0 for k<0

The code is punctured so that the following 80 coded bits:

{C(8*j+3), C(8*j+7) for j = 0, 1, ..., 21

C(8*j+3), C(8*j+5), C(8*j+7) for j = 22, 23, ..., 32)}

and the bits C(1), C(265) and C(267) are not transmitted.

As a result, there is a block of 188 coded and punctured bits, P(0)...P(187) which is appended to the bits of an in-band signaling in c.

c(k+4) = P(k) for k = 0, 1, ..., 187.

Finally, 36 Class-2 bits are appended to c

c(192+k) = d(123+k) for k = 0, 1, ..., 35.

CH9-HS:

The 262 coded bits {C(0)...C(261)}

C(2k) = u(k)

C(2k+1) = u(k)+u(k-2)+u(k-4)+u(k-5)+C(2k-1)+C(2k-3)+ C(2k-5)+C(2k-9)

for k = 0, 1, ..., 130; u(k) = 0 for k<0; C(k) = 0 for k<0 are punctured so that 66 coded bits:

{C(16*j+3), C(16*j+7), C(16*j+11) for j = 0, 1, ..., 7

C(16*j+3), C(16*j+7), C(16*j+11), C(16*j+15) for j = 8, 9, ..., 15)} and the bits C(1), C(221), C(229), C(237), C(245), C(249), C(253), C(257), C(259) and C(261) are not transmitted.

A block of 196 coded and punctured bits, P(0)...P(195) is appended to the bits of the in-band signaling in c:

c(k+4) = P(k) for k = 0, 1, ..., 195.

Finally, 28 Class-2 bits are appended to c:

c(200+k) = d(120+k) for k = 0, 1, ..., 27.

CH10-HS:

The 242 coded bits {C(0)...C(241)}:

C(2k) = u(k)

C(2k+1) = u(k)+u(k-2)+u(k-4)+u(k-5)+C(2k-1)+C(2k-3)+ C(2k-5)+C(2k-9)

for k = 0, 1, ..., 106; u(k) = 0 for k<0; C(k) = 0 for k<0

are punctured so that 42 coded bits:

{C(8*j+3) for j = 0, 1, ..., 21

C(8*j+3), C(8*j+7) for j = 22, 23, ..., 29)} and the bits C(1), C(233), C(237) and C(241) are not transmitted.

A block of 200 coded and punctured bits, P(0)...P(199) is appended to the bits of the in-band signaling in c:

c(k+4) = P(k) for k = 0, 1, ..., 199.

Finally, 24 Class-2 bits are appended to c:

c(204+k) = d(110+k) for k = 0, 1, ..., 23.

CH11-HS:

The 226 coded bits {C(0)...C(225)}:

C(2k) = u(k)

C(2k+1) = u(k)+u(k-2)+u(k-4)+u(k-5)+C(2k-1)+C(2k-3)+ C(2k-5)+C(2k-9)

for k = 0, 1, ..., 112; u(k) = 0 for k<0; C(k) = 0 for k<0

are punctured so that 18 coded bits:

{C(28*j+15) for j = 0, 1, ..., 7} and bits C(1), C(3), C(7), C(197), C(213), C(215), C(217), C(221), C(223) and C(225) are not transmitted.

A block of 208 coded and punctured bits, P(0)...P(207) is appended to the bits of the in-band signaling in c:

c(k+4) = P(k) for k = 0, 1, ..., 207.

Finally, 16 Class-2 bits are appended to c:

c(212+k) = d(96+k) for k = 0, 1, ..., 15.

CH12-HS:

The 309 coded bits {C(0)...C(308)}:

C(3k) = u(k)

C(3k+1) = u(k)+u(k-1)+u(k-2)+u(k-3)+u(k-6)+C(3k-5)+ C(3k-8)+C(3k-14)+C(3k-17)

C(3k+2) = u(k)+u(k-1)+u(k-4)+u(k-6)+C(3k-4)+C(3k-7)+ C(3k-11)+C(3k-16)

for k = 0, 1, ..., 102; u(k) = 0 for k<0; C(k) = 0 for k<0

are punctured so that 97 coded bits: {C(12*j+5), C(12*j+8), C(12*j+11) for j = 0, 1, ..., 15 C(12*j+2), C(12*j+5), C(12*j+8), C(12*j+11) for j = 16, 17, ..., 24) and bits C(1), C(2), C(4), C(7), C(292), C(292), C(295), C(298), C(301), C(302), C(304), C(305), C(307) and C(308) are not transmitted.

A block of 212 coded and punctured bits, P(0)...P(211) is appended to the bits of the in-band signaling in c:

c(k+4) = P(k) for k = 0, 1, ..., 211.

Finally, 12 Class-2 bits are appended to c:

c(216+k) = d(91+k) for k = 0, 1, ..., 11.

CH13-HS:

The 285 coded bits {C(0)...C(284)}:

C(3k) = u(k)

C(3k+1) = u(k)+u(k-1)+u(k-2)+u(k-3)+u(k-6)+C(3k-5)+ C(3k-8)+C(3k-14)+C(3k-17)

C(3k+2) = u(k)+u(k-1)+u(k-4)+u(k-6)+C(3k-4)+C(3k-7)+ C(3k-11)+C(3k-16)

for k = 0, 1, ..., 94; u(k) = 0 for k<0; C(k) = 0 for k<0

are punctured so that 73 coded bits:
{C(12*j+5), C(12*j+11) for j = 0, 1, ..., 11
C(12*j+5), C(12*j+8), C(12*j+11) for j = 12, 13, ..., 22} and bits C(1), C(2), C(4), C(7), C(8), C(14), C(242), C(254), C(266), C(274), C(277), C(278), C(280), C(281), C(283) and C(284) are not transmitted.

A block of 212 coded and punctured bits, P(0)...P(211) is appended to the bits of the in-band signaling in c:

c(k+4) = P(k) for k = 0, 1, ..., 211.

Finally, 12 Class-2 bits are appended to c:

c(216+k) = d(91+k) for k = 0, 1, ..., 11.

The polynomials of the systematic recursive code (G13 to G17) in the AMR (see Figure 5) shown were used for two reasons:

- they have optimum characteristics for the puncturing, i.e. the adaptation of the data rate to the transmission rate of the radio channel, and
- numerator or denominator polynomial are in each case also the polynomial used in the original AMR channel coding proposal (see Tdoc SMG 147/98). The necessary changes are thus minimum compared with the original proposal.

The polynomials used hitherto for voice, data and signaling information in the GSM system can also be used for the AMR channel coder with negligible restrictions in the performance. This can be done instead of the polynomials described above or as a complete alternative channel coding arrangement. The advantage lies in that the compatibility is extended further since in some cases older pre-existing hardware components in the channel decoder only allow the previous GSM polynomials to be used.

Figure 6 shows a base station BS in which, in the reception case, signals received via an antenna A are amplified in a receiver, filtered, converted to baseband and digitized.

This is followed by channel decoding (step 1), which can be done with decoding devices installed in existing base stations BS, i.e. the circuit technology can remain unchanged. This is followed by post-processing (step 2) of the decoded data which is implemented as a program. This post-processing consists of convolutional coding at a rate of 1 with the denominator polynomial of the respective rate.

As a result, this post-processing is of little complexity and is performed, for example, by an additional program in a DSP (digital signal processor).

Referring, e.g. to the rate CH0-FS, this means that the block with 255 bits at the output of the decoder must be coded with the polynomial:

$$G(D) = (1 + D + D^2 + D^3 + D^5)$$

in order to obtain the 255 original bits. The number of data bits remains constant, i.e. a current data bit at the input of this post-processing yields exactly one original bit with the aid of past input bits.

The coding and decoding methods described can be used both in base stations BS and in mobile stations MS.

Patent claims

1. A method for channel coding in a GSM mobile radio system, in which a channel coding which uses recursive system codes with a numerator polynomial and a denominator polynomial is performed at the transmitting end for the transmission via a radio interface between a base station (BS) and a subscriber station (MS).

2. The method as claimed in claim 1, in which a nonrecursive channel decoding is performed at the receiving end.

3. The method as claimed in claim 2, in which, after channel decoding with the numerator polynomial, post-processing is performed on the basis of the denominator polynomial.

4. The method as claimed in claim 3, in which the post-processing is performed by programming means.

5. The method as claimed in one of the previous claims, in which a priori knowledge is obtained from previous decoding at the receiving end and this a priori knowledge is used in subsequent channel decoding.

6. The method as claimed in one of the previous claims, in which the channel decoding is completely switched off in a subscriber station (MS) and thereafter the transmitted systematic data bits which are not channel coded are used.

7. The method as claimed in one of the previous claims, in which a transmission quality is determined during a channel estimation, and

the channel decoding is switched on or off in dependence on the transmission quality.

8. The method as claimed in one of the previous claims, in which the recursive systematic codes are used in an adaptive multirate coder, a coder being selected in accordance with the transmission conditions.

9. The method as claimed in one of the previous claims, in which, of the two polynomials of the recursive systematic codes, at least one polynomial of a nonrecursive systematic code previously used in the GSM mobile radio system is used.

10. The base station (BS) for a GSM mobile radio system which performs, for the transmission via a radio interface to a subscriber station (MS), a channel coding which uses recursive systematic codes comprising a numerator polynomial and a denominator polynomial.

11. A subscriber station (MS) for a GSM mobile radio system which performs, for the transmission via a radio interface to a base station (BS), a channel coding which uses recursive systematic codes comprising a numerator polynomial and a denominator polynomial.

12. The subscriber station (MS) as claimed in claim 11, comprising a channel decoder which can be switched off.

13. The subscriber station (MS) as claimed in claim 12, comprising a channel decoder which, in the switched-off state, forwards the transmitted data which are not channel coded.

Key to figures

Figure 1:
Stand der Technik = Prior art

Figure 2:
Nonsystematic nonrecursive code with memory 4 and rate 1/2 analogously to GSM/TCHFS

Figure 3:
Identical recursive systematic convolutional code with memory 4 and rate 1/2

Figure 4:

1 In-band data
2 Voice frames
3 Sorting
4 Class 2
5 Block code
6 Convolutional code
7 or
8 Reordering and distribution
9 Diagonal interleaving
in: 4 blocks
out: block pairs
10 Encryption

Figure 5:
Polynomials used in different channels in the GSM mobile radio system

1 User data channel, adaptive multirate coding, full rate
2 User data channel, adaptive multirate coding, half rate

Figure 6:

1 Receiver
2 Channel decoder
3 Post-processing

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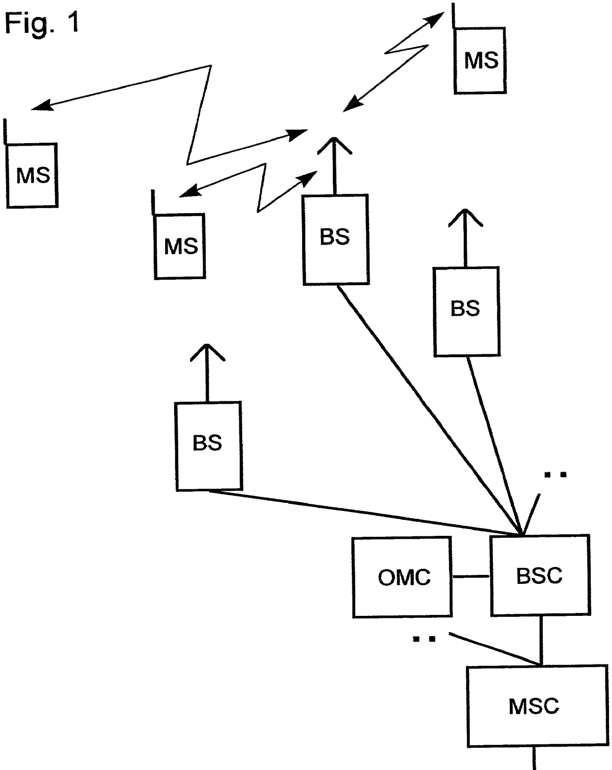
Fig. 1



(Stand der Technik)

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Fig. 2

nichtsytematischer nichtrekursiver Kode mit Gedächtnis 4 und Rate 1/2, analog zu GSM/TCHFS

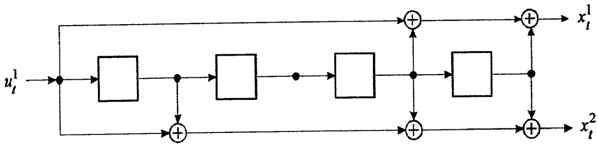


Fig. 3

identischer rekursiver systematischer Faltungscode mit Gedächtnis 4 und Rate 1/2

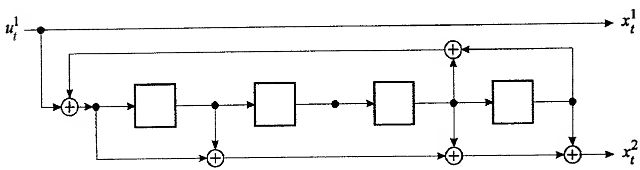
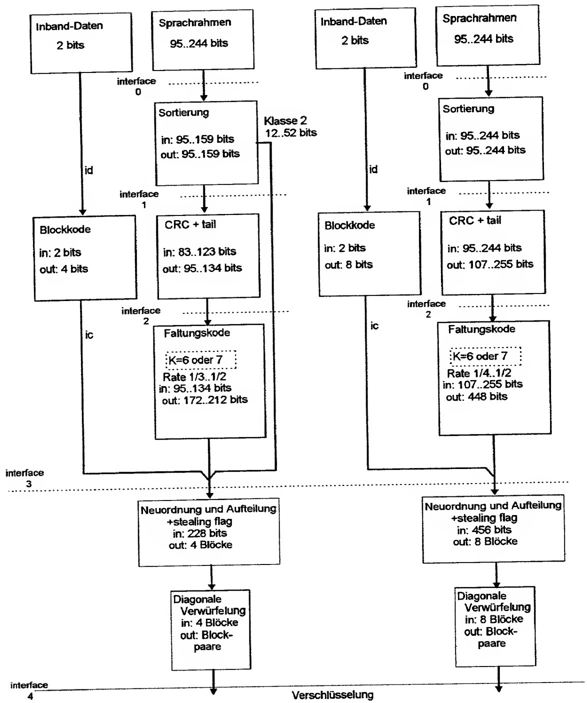


Fig. 4



Inband-Daten
2 bits
Sprachrahmen
95..244 bits
interface
0
Sortierung
in: 95..159 bits
out: 95..159 bits
Klasse 2
12..52 bits
id
interface
1
Blockkode
in: 2 bits
out: 4 bits
CRC + tail
in: 83..123 bits
out: 95..134 bits
interface
2
ic
Faltungskode
K=6 oder 7
Rate 1/3..1/2
in: 95..134 bits
out: 172..212 bits
Inband-Daten
2 bits
Sprachrahmen
95..244 bits
interface
0
Sortierung
in: 95..244 bits
out: 95..244 bits
id
interface
1
Blockkode
in: 2 bits
out: 8 bits
CRC + tail
in: 95..244 bits
out: 107..255 bits
interface
2
ic
Faltungskode
K=6 oder 7
Rate 1/4..1/2
in: 107..255 bits
out: 448 bits
interface
3
Neuordnung und Aufteilung
+stealing flag
in: 228 bits
out: 4 Blöcke
Neuordnung und Aufteilung
+stealing flag
in: 456 bits
out: 8 Blöcke
Diagonale
Verwürfelung
in: 4 Blöcke
out: Block-
paare
Diagonale
Verwürfelung
in: 8 Blöcke
out: Block-
paare
interface
4
Verschlüsselung

Fig. 5

im GSM-Mobilfunksystem in unterschiedlichen Kanälen verwendete Polynome

$G0 = 1 + D^3 + D^4$	
$G1 = 1 + D + D^3 + D^4$	
$G2 = 1 + D^2 + D^4$	
$G3 = 1 + D + D^2 + D^3 + D^4$	
$G4 = 1 + D^2 + D^3 + D^5 + D^6$	TCH/AMR-FS, TCH/AMR-HS
$G5 = 1 + D + D^4 + D^6$	TCH/AMR-FS, TCH/AMR-HS
$G6 = 1 + D + D^2 + D^3 + D^4 + D^6$	
$G7 = 1 + D + D^2 + D^3 + D^5$	TCH/AMR-FS, TCH/AMR-HS
$G8 = 1 + D^2 + D^4 + D^5$	TCH/AMR-FS, TCH/AMR-HS
$G9 = 1 + D^3 + D^4 + D^5$	TCH/AMR-FS
$G10 = 1 + D + D^2 + D^3 + D^6$	TCH/AMR-FS, TCH/AMR-HS
$G11 = 1 + D^2 + D^3 + D^4 + D^5 + D^6$	TCH/AMR-FS
$G12 = 1$	TCH/AMR-FS, TCH/AMR-HS
$G13 = G8 / G7$	TCH/AMR-FS, TCH/AMR-HS
$G14 = G9 / G7$	TCH/AMR-FS
$G15 = G10 / G4$	TCH/AMR-FS, TCH/AMR-HS
$G16 = G5 / G4$	TCH/AMR-FS, TCH/AMR-HS
$G17 = G11 / G4$	TCH/AMR-FS

TCH/AMR-FS	Nutzdatenkanal, adaptive Multiratenkodierung, Fullrate
TCH/AMR-HS	Nutzdatenkanal, adaptive Multiratenkodierung, Halfrate

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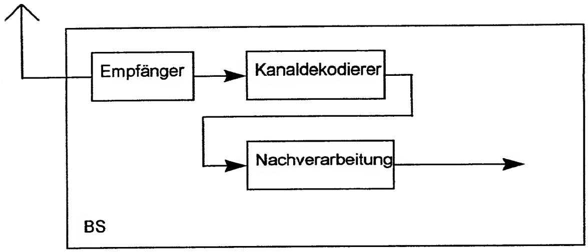
Fig. 6



Declaration and Power of Attorney For Patent Application
Erklärung Für Patentanmeldungen Mit Vollmacht
German Language Declaration

Als nachstehend benannter Erfinder erkläre ich hiermit an Eides Statt:

dass mein Wohnsitz, meine Postanschrift, und meine Staatsangehörigkeit den im Nachstehenden nach meinem Namen aufgeführten Angaben entsprechen,

dass ich, nach bestem Wissen der ursprüngliche, erste und alleinige Erfinder (falls nachstehend nur ein Name angegeben ist) oder ein ursprünglicher, erster und Miterfinder (falls nachstehend mehrere Namen aufgeführt sind) des Gegenstandes bin, für den dieser Antrag gestellt wird und für den ein Patent beantragt wird für die Erfindung mit dem Titel:

Verfahren, Basisstation und Teilnehmerstation zur Kanalkodierung in einem GSM-Mobilfunksystem

deren Beschreibung

(zutreffendes ankreuzen)
☐ hier beigefügt ist.
☒ am 19.11.1999 als
PCT internationale Anmeldung
PCT Anmeldungsnummer PCT/DE99/03698
eingereicht wurde und am ________
abgeändert wurde (falls tatsächlich abgeändert).

Ich bestätige hiermit, dass ich den Inhalt der obigen Patentanmeldung einschliesslich der Ansprüche durchgesehen und verstanden habe, die eventuell durch einen Zusatzantrag wie oben erwähnt abgeändert wurde.

Ich erkenne meine Pflicht zur Offenbarung irgendwelcher Informationen, die für die Prüfung der vorliegenden Anmeldung in Einklang mit Absatz 37, Bundesgesetzbuch, Paragraph 1.56(a) von Wichtigkeit sind, an.

Ich beanspruche hiermit ausländische Prioritätsvorteile gemäss Abschnitt 35 der Zivilprozessordnung der Vereinigten Staaten, Paragraph 119 aller unten angegebenen Auslandsanmeldungen für ein Patent oder eine Erfindersurkunde, und habe auch alle Auslandsanmeldungen für ein Patent oder eine Erfindersurkunde nachstehend gekennzeichnet, die ein Anmeldedatum haben, das vor dem Anmeldedatum der Anmeldung liegt, für die Priorität beansprucht wird.

As a below named inventor, I hereby declare that:

My residence, post office address and citizenship are as stated below next to my name,

I believe I am the original, first and sole inventor (if only one name is listed below) or an original, first and joint inventor (if plural names are listed below) of the subject matter which is claimed and for which a patent is sought on the invention entitled

Method, base station and subscriber station for channel coding in a gsm mobile radiotelephone system

the specification of which

(check one)
☐ is attached hereto.
☒ was filed on 19.11.1999 as
PCT international application
PCT Application No. PCT/DE99/03698
and was amended on ________
(if applicable)

I hereby state that I have reviewed and understand the contents of the above identified specification, including the claims as amended by any amendment referred to above.

I acknowledge the duty to disclose information which is material to the examination of this application in accordance with Title 37, Code of Federal Regulations, §1.56(a).

I hereby claim foreign priority benefits under Title 35, United States Code, §119 of any foreign application(s) for patent or inventor's certificate listed below and have also identified below any foreign application for patent or inventor's certificate having a filing date before that of the application on which priority is claimed:

IDNR: 2590 / V: 99-1.00 / B:Val
Form PTO-FB-240 (8-83)
Patent and Trademark Office-U.S. DEPARTMENT OF COMMERCE

German Language Declaration

Prior foreign appplications
Priorität beansprucht

Priority Claimed

19853443.4 (Number) (Nummer)	DE (Country) (Land)	19.11.1998 (Day Month Year Filed) (Tag Monat Jahr eingereicht)	☒ Yes Ja	☐ No Nein
(Number) (Nummer)	(Country) (Land)	(Day Month Year Filed) (Tag Monat Jahr eingereicht)	☐ Yes Ja	☐ No Nein
(Number) (Nummer)	(Country) (Land)	(Day Month Year Filed) (Tag Monat Jahr eingereicht)	☐ Yes Ja	☐ No Nein

Ich beanspruche hiermit gemäss Absatz 35 der Zivilprozessordnung der Vereinigten Staaten, Paragraph 120, den Vorzug aller unten aufgeführten Anmeldungen und falls der Gegenstand aus jedem Anspruch dieser Anmeldung nicht in einer früheren amerikanischen Patentanmeldung laut dem ersten Paragraphen des Absatzes 35 der Zivilprozeßordnung der Vereinigten Staaten, Paragraph 122 offenbart ist, erkenne ich gemäss Absatz 37, Bundesgesetzbuch, Paragraph 1.56(a) meine Pflicht zur Offenbarung von Informationen an, die zwischen dem Anmeldedatum der früheren Anmeldung und dem nationalen oder PCT internationalen Anmeldedatum dieser Anmeldung bekannt geworden sind.

I hereby claim the benefit under Title 35. United States Code. §120 of any United States application(s) listed below and, insofar as the subject matter of each of the claims of this application is not disclosed in the prior United States application in the manner provided by the first paragraph of Title 35, United States Code, §122, I acknowledge the duty to disclose material information as defined in Title 37, Code of Federal Regulations, §1.56(a) which occured between the filing date of the prior application and the national or PCT international filing date of this application.

PCT/DE99/03698 (Application Serial No.) (Anmeldeseriennummer)	19.11.1999 (Filing Date D, M, Y) (Anmeldedatum T, M, J)	(Status) (patentiert, anhängig, aufgegeben)	(Status) (patented, pending, abandoned)
(Application Serial No.) (Anmeldeseriennummer)	(Filing Date D,M,Y) (Anmeldedatum T, M; J)	(Status) (patentiert, anhängig, aufgeben)	(Status) (patented, pending, abandoned)

Ich erkläre hiermit, dass alle von mir in der vorliegenden Erklärung gemachten Angaben nach meinem besten Wissen und Gewissen der vollen Wahrheit entsprechen, und dass ich diese eidesstattliche Erklärung in Kenntnis dessen abgebe, dass wissentlich und vorsätzlich falsche Angaben gemäss Paragraph 1001, Absatz 18 der Zivilprozessordnung der Vereinigten Staaten von Amerika mit Geldstrafe belegt und/oder Gefängnis bestraft werden koennen, und dass derartig wissentlich und vorsätzlich falsche Angaben die Gültigkeit der vorliegenden Patentanmeldung oder eines darauf erteilten Patentes gefährden können.

I hereby declare that all statements made herein of my own knowledge are true and that all statements made on information and belief are believed to be true, and further that these statements were made with the knowledge that willful false statements and the like so made are punishable by fine or imprisonment, or both, under Section 1001 of Title 18 of the United States Code and that such willful false statements may jeopardize the validity of the application or any patent issued thereon.

German Language Declaration

VERTRETUNGSVOLLMACHT: Als benannter Erfinder beauftrage ich hiermit den nachstehend benannten Patentanwalt (oder die nachstehend benannten Patentanwälte) und/oder Patent-Agenten mit der Verfolgung der vorliegenden Patentanmeldung sowie mit der Abwicklung aller damit verbundenen Geschäfte vor dem Patent- und Warenzeichenamt: *(Name und Registrationsnummer anführen)*

POWER OF ATTORNEY: As a named inventor, I hereby appoint the following attorney(s) and/or agent(s) to prosecute this application and transact all business in the Patent and Trademark Office connected therewith. *(list name and registration number)*

29177 And I hereby appoint
PATENT & TRADEMARK OFFICE

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Postanschrift	Post Office Address

(Bitte entsprechende Informationen und Unterschriften im Falle von dritten und weiteren Miterfindern angeben).

(Supply similar information and signature for third and subsequent joint inventors).